Software Utilities Manual

Version 3.80

Catalog Number: LTRT-00302

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This manual provides additional information for the following AudioCodes products:

MediaPack Series:	MP-100, MP-104, MP-108, MP-124, MP-200
TrunkPack Series:	TP-240, TP-400, TP-610, TP-610/ATM
TrunkPack Module Series:	TPM-200, TPM-200/RDK, TPM-800, TPM-800/RDK
IPmedia Series:	IPM-610

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Each abbreviation, unless widely used, is spelled out in full when first used. Only industry-standard terms are used throughout this manual. The $ symbol indicates hexadecimal notation.

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Date Published: Nov-30-2001 Date Printed: Dec-09-2001

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Appendix A- BootP & TFTP Server Configuration Tool

This section describes the features, specifications, installation/downloading and functionalities of the BootP & TFTP Server Configuration Tool

A.1 Introduction

The **BootP & TFTP Server Configuration Tool** enables easy configuration and provisioning of AudioCodes products. It contains BootP and TFTP servers with specific adaptations to AudioCodes' requirements.

A.1.1 Key Features

- Internal BootP server supporting hundreds of entities.
- Internal TFTP server.
- Contains all required data for AudioCodes products in predefined format.
- Provides a TFTP server address, enabling network separation of TFTP and BootP servers.
- Tools to backup and restore the local database.
- Templates.
- User-defined names for each entity.
- Option for changing MAC address.
- Protection against entering faulty information.
- Remote reset (for this version and above).
- Unicast BootP respond.
- User-initiated BootP respond, for remote provisioning over WAN.
- Filtered display of BootP requests.
- Location of other BootP servers that contain the same MAC entity.
- Common log window for both BootP and TFTP sessions.
- Works with Windows™ 98, Windows™ NT, Windows™ 2000.

A.1.2 Specifications

BootP standards:	RFC 951 and RFC 1542
TFTP standards:	RFC 1350 and RFC 906
Operating System:	Windows™ 98, Windows™ NT and Windows™ 2000
Max number of MAC entries:	200

BootP Fields:

- Hardware address (MAC): 12 hex digits.
- IP address.
- Subnet.
- Default Gateway.
- TFTP server IP; (Using the TFTP server IP field enables the download of a software image from a different Host).
- Boot File.
- .ini File.
- Call Agent IP.
- New MAC (optional).

Screens:

- Logging screen.
- Preferences screen.
- Client Configuration screen.
- Template definition screen.

A.1.3 BootP/TFTP Server Installation

The BootP/TFTP Server Configuration Tool can be installed on a PC from the AudioCodes Software & Documentation CD:

TrunkPack Boards (TP)	Compact disk Catalog Number: LSTC-00001,
IPmedia Platforms (IPM)	Compact disk Catalog Number: LSTC-00002
TrunkPack Modules (TPM)	Compact disk Catalog Number: LSTC-00003

MediaPack Series (MP) Compact disk Catalog Number: LSTC-00005

To install the BootP/TFTP Server Configuration Tool Demo, unzip the TP3.80.exe file and navigate to the BootP *.exe file. The installation procedure is facilitated by prompts. After completing the procedure, open Start>Programs>BootP; the BootP/TFTP Server main screen is displayed.

A.1.4 Logging Screen

The AudioCodes BootP/TFTP Server main screen (Figure A-1 on page 15) includes the Log line, printed per BootP request with the following parameters:

- Hardware (MAC) address.
- Status (found or not found in cache).
- Date and Time.
- Assigned IP address (if found).
- Client name.

Clicking on Log line shows all BootP reply parameters or enables entry to a new entity.

A.1.5 Preferences Window

The Preferences Window (Figure A-2 on page 17) is used to define BootP and TFTP configuration parameters:

- TFTP directory.
- *ini* File Mask.
- Boot File Mask.
- TFTP timeout and number of retransmissions.
- BootP replay type (broadcast or unicast).
- BootP ARP mode (dynamic or static).
- Number of initiated BootP replies (send after remote reset), optionally used when the Media Gateway (MP-100, MP-108, MP-124 and MP-200 or TP board/module) is installed behind the firewall that blocks BootP broadcast requests.

A.1.6 Client Configuration Window

The Client Configuration Window (Figure A-3 on page 19) shows:

- All client entities
- MAC
- Name
- IP per entity.

Using this screen, the User can:

- Add a new entry
- Delete an existing entry
- Modify an existing entry
- Test a selected client for finding all BootP servers that respond to a BootP request with a specific MAC address.

If a template is selected, any parameter can be entered manually or copied from the selected template, by marking the checkbox on the right side of the parameter. Usually, only an IP address is entered manually, while other parameters are copied from the template.

A.1.7 Template Window

The Template Window (Figure A-4 on page 21) enables the User to add, modify, or delete templates.

The template includes:

- Subnet.
- Gateway, TFTP server.
- BootFile.
- *ini* file.
- Call Agent fields.
- Server IP.

A.2 Window Details

A.2.1 Main Window

Figure A-1: Main Window

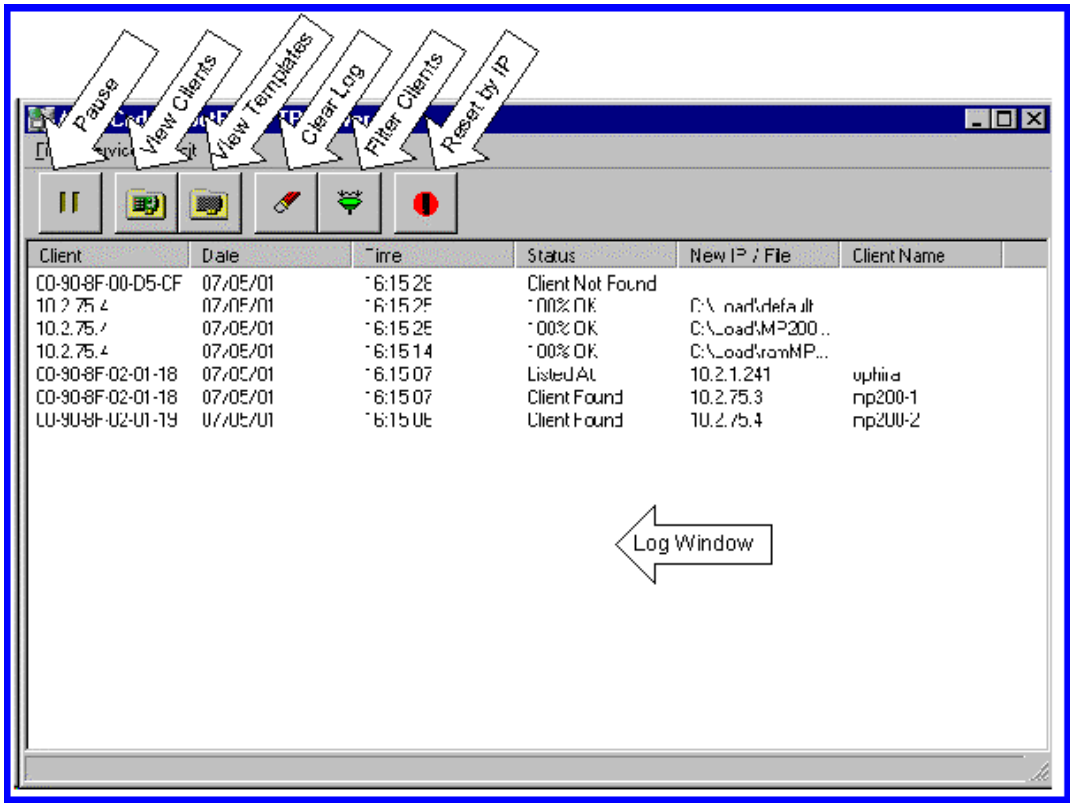


Figure A-1 is the main window of the program. It features several controls, listed below:

- **Program State** - With this button, the User can pause the program. When the program has paused, no replies to BootP requests are sent.
- **View Clients Button** - This button opens up the Clients Configuration window.

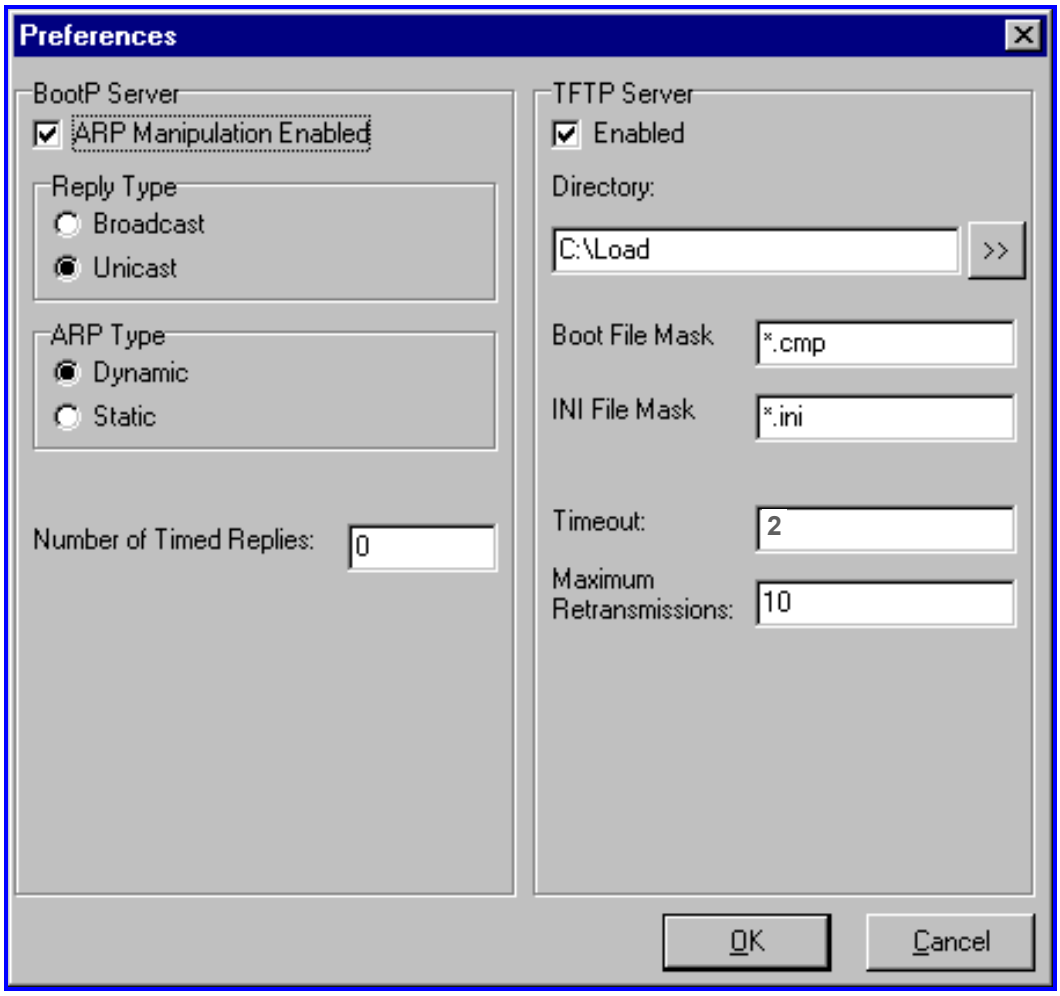
- **View Templates Button** - This button opens up the Templates Configuration window.
- **Clear Log Button** - This buttons clears the log.
- **Filter Unknown Clients Button** - This button filters all BootP requests that are not in the client configuration list.
- **Reset Button** - This button opens up a dialog box, where the User can enter an IP of a client. The program sends a reset command to that client.
- **Edit/Preferences** - Selecting "Preferences" from the Edit menu opens the Preferences window for defining BootP and TFTP parameters.
- **Log Window** - All BootP requests and TFTP sessions are displayed, including the time and date of the request. In addition, the response type is also displayed:
  - "**Client Not Found**"
  - "**Client Found**"
  - "**Client's Mac Changed**"
  - **"Client Disabled"**
  - **"Listed at"** (when using the "test selected clients" button).
  - For TFTP session, **File name and Download status** are displayed.

- **Pop-Up Menu** - When the User right-clicks on a line in the log window, the pop-up menu opens. In this menu there two options:
  - **Reset**: When this option is selected, the program searches the database for the selected MAC. When the client is found, the program adds the client's MAC to the ARP table, and then sends a reset command to the client. Note that by performing the remote reset this way, the User doesn't have to know the current IP of the client. **To perform this, the User must have administrator privileges**, or else an error message appears.
  - **View Client**: This option is the same as double-clicking on a line. When selected, the Clients Window opens. If the Client's MAC is found in the database, it is focused. If not, a new client is added, with the MAC filled out. The User must only fill in the remaining fields.

A.2.2 Preferences Window

In the Preferences window, Figure A-2, BootP and TFTP configuration parameters are defined.

Figure A-2: Preferences Window



For the TFTP server, the User can configure a TFTP directory and a value for TFTP Timeout and Maximum Retransmissions. Set these values to 2 and 10 as shown above.

The TFTP server can be disabled by clearing the Enable checkbox.

In the BootP section, the User can select ARP mode: Dynamic or Static, and reply type: Broadcast or Unicast. For a typical application, use Dynamic ARP mode and Unicast, as shown above.

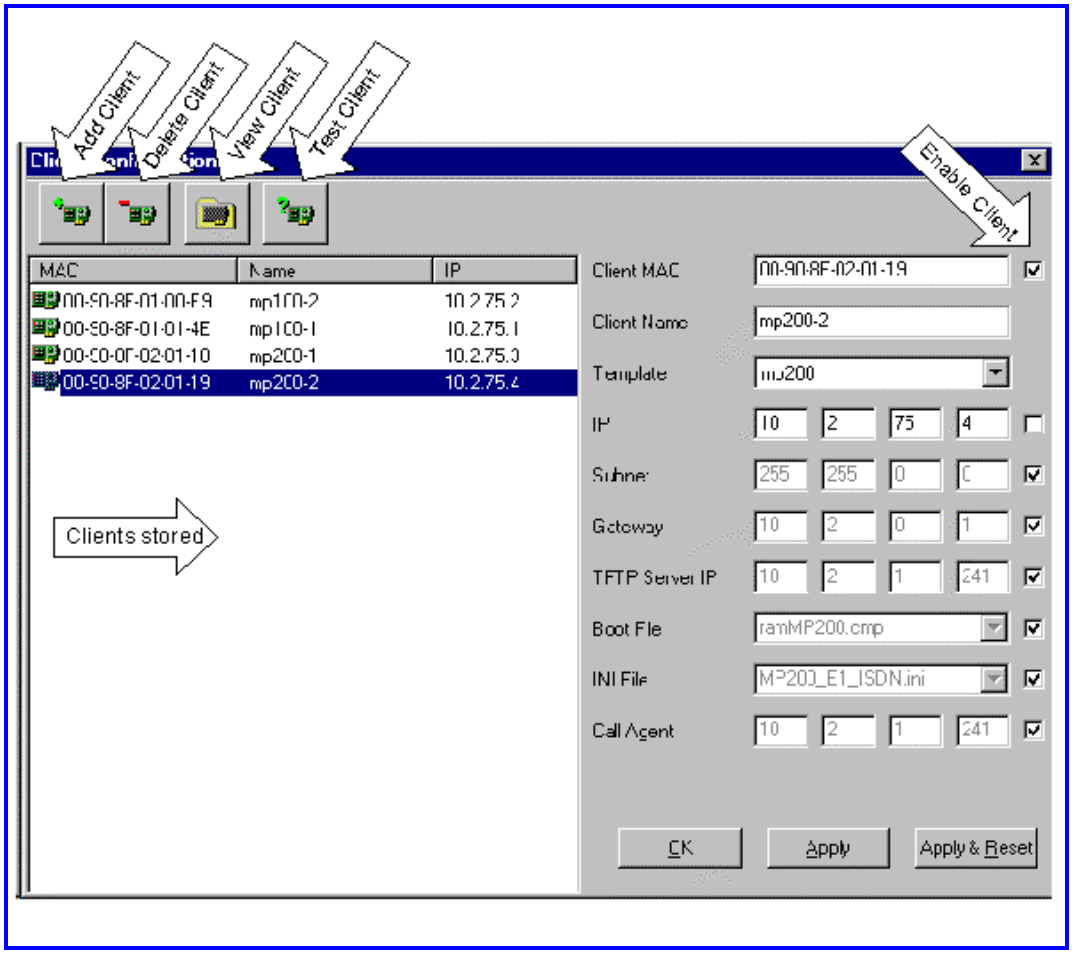
This option requires the **User to have administrator privileges** otherwise an error message appears**.** If you don't have administrator privileges**, uncheck** the ARP Manipulation Enabled checkbox in the Preferences Window, Figure A-2.

The "Number of Timed Replies" (the number of initiated timed BootP replies) can be used when an MP-1xx or MP-200 (or TP board/module) is installed behind a Firewall that blocks BootP broadcast requests. In a typical application, this feature can be disabled by entering "0" in this field. When selected, several BootP replies are sent to the MP-1xx or MP-200 (or to the TP board/module) immediately after the remote reset command.

A.2.3 Client Configuration Window

Figure A-3 is the Client Configuration Window in which clients are added and defined.

Figure A-3: Client Configuration Window



In the left pane of the window is the client list. By clicking on a client in this list, the following parameters for this client are displayed on the right side of the window:

- **Client MAC**: This is the MAC address of the client. When the User edits the MAC, a new client is added, with the same parameters as the previous client.

  The client can be **disabled by un-checking** the check box on the right side of the Client MAC, causing the BootP server not to reply to the BootP request. The client can be **enabled by checking** the check box. Click on the 'Apply' button **each time** the client enable check box is checked or unchecked.

- **Client Name**: Free text for client description.
- **Template**: The template to be used for this client. When a template is selected, its parameters override all of the previous parameters.
- **IP, Subnet, Gateway**: Normal IP parameters.
- **TFTP Server IP**: The IP of the TFTP Server.
- **Boot File, *ini* File**: The files to request from the TFTP server.
- **Call Agent**: The IP of the MGCP Call Agent that will be controlling the gateway.

Note the seven check boxes to the right of the parameters. These enable the User to assign only the selected fields from the template. The rest can be unique for each client. When the field is assigned a value from the selected template, the field is grayed out.

After performing changes, press "Apply" to save them. By clicking on "Apply & Reset", the program saves the changes to the database, performs a remote reset to the client by adding the client's MAC to the ARP table, and then sends out a reset command. This option works **only if "ARP Manipulation Enabled"** checkbox in the "Preferences" window is **checked** (in Figure A-2 on page 17) otherwise an error message appears. It requires the User to have **administrator privileges**. The remote reset is supported for software in this version and up.

When adding a new client, click on the "Add Client" button; a client with blank parameters is displayed. After filling out the parameters, click on "Apply"; the client is added.

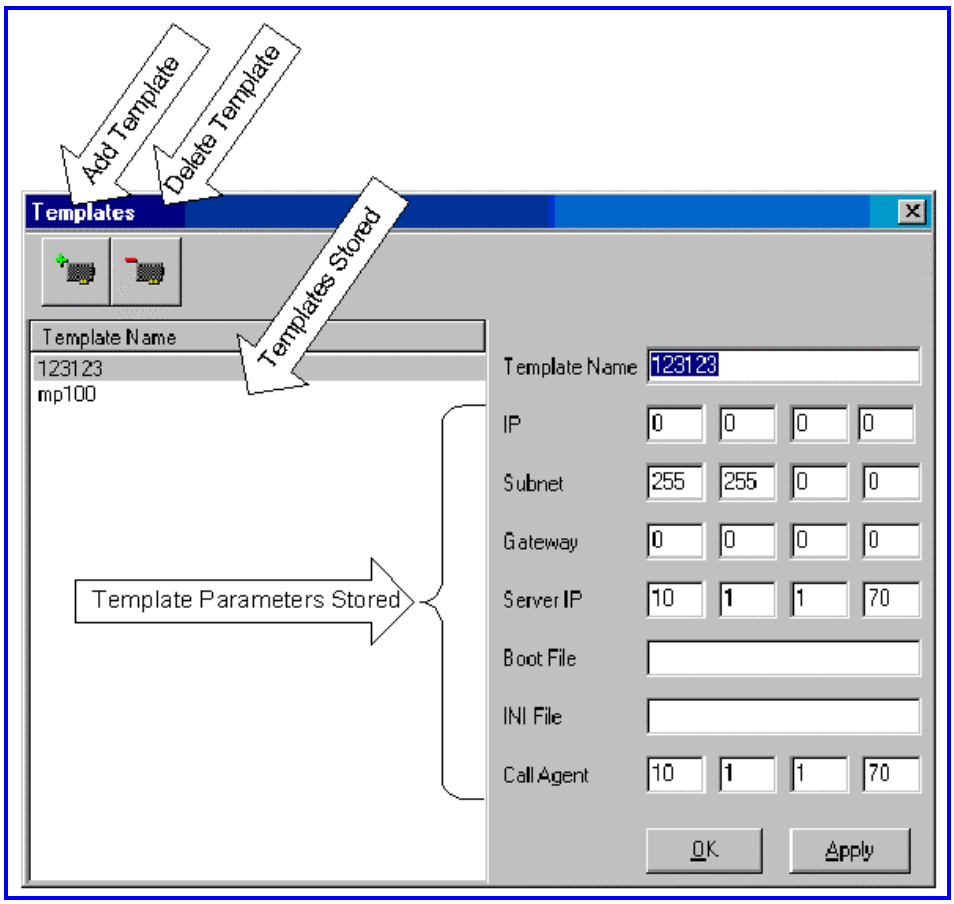
To find out if there is another BootP server on the net that contains a client with the same MAC address, click on "Test Selected Clients"; in the log screen, view the IP addresses of all BootP servers that contain the same MAC

address in the status 'Listed At'. In normal operation, BootP client MAC address should be listed only on a single BootP server. If the MAC address is listed in multiple BootP servers, it must be removed from other BootP servers.

A.2.4 Templates Window

Figure A-4 shows the Templates window, which provides a fast way to configure a number of clients that have the same parameters (except for the IP address). To use the Templates window, create a template, and then apply the template to the client by selecting it.

Figure A-4: Templates Window



Appendix B- DHCP Server Configuration

This section describes the DHCP Server Configuration.

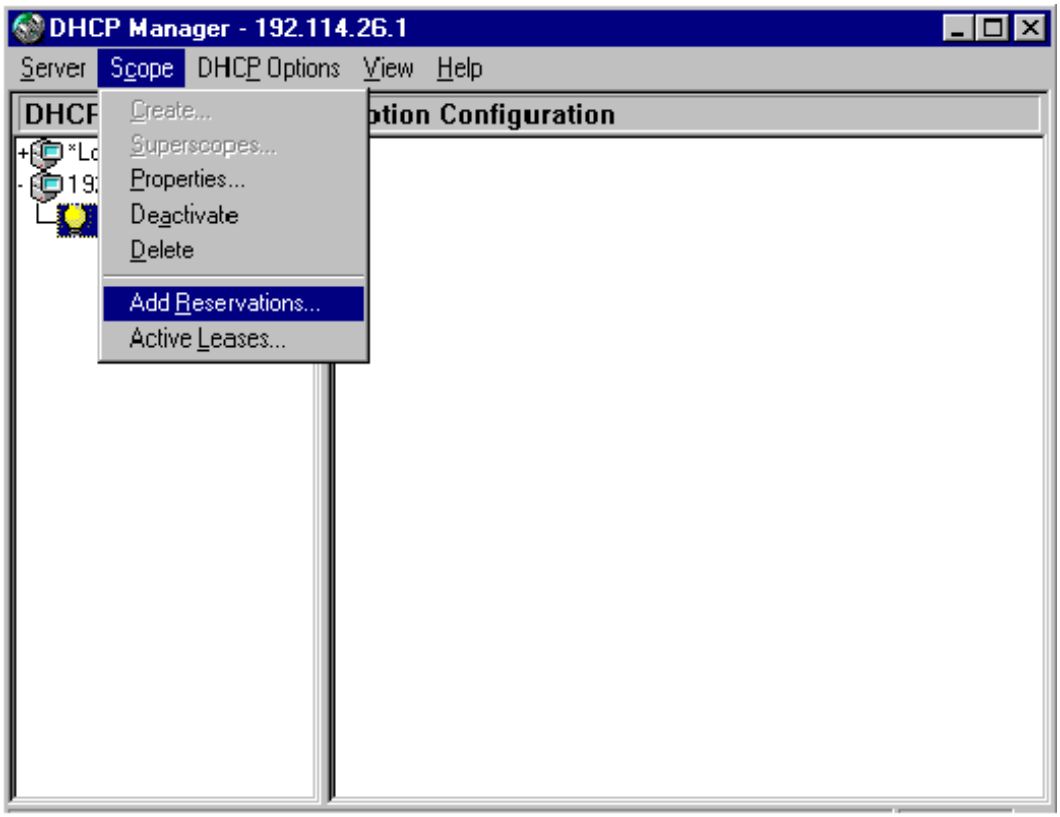
B.1 Configuration

Note: The User needs to install Windows™ NT4 service pack 4 after enabling the DHCP server service on the NT server. This is required for correct operation with BootP clients.

➢ To configure the DHCP server take the next 16 steps:

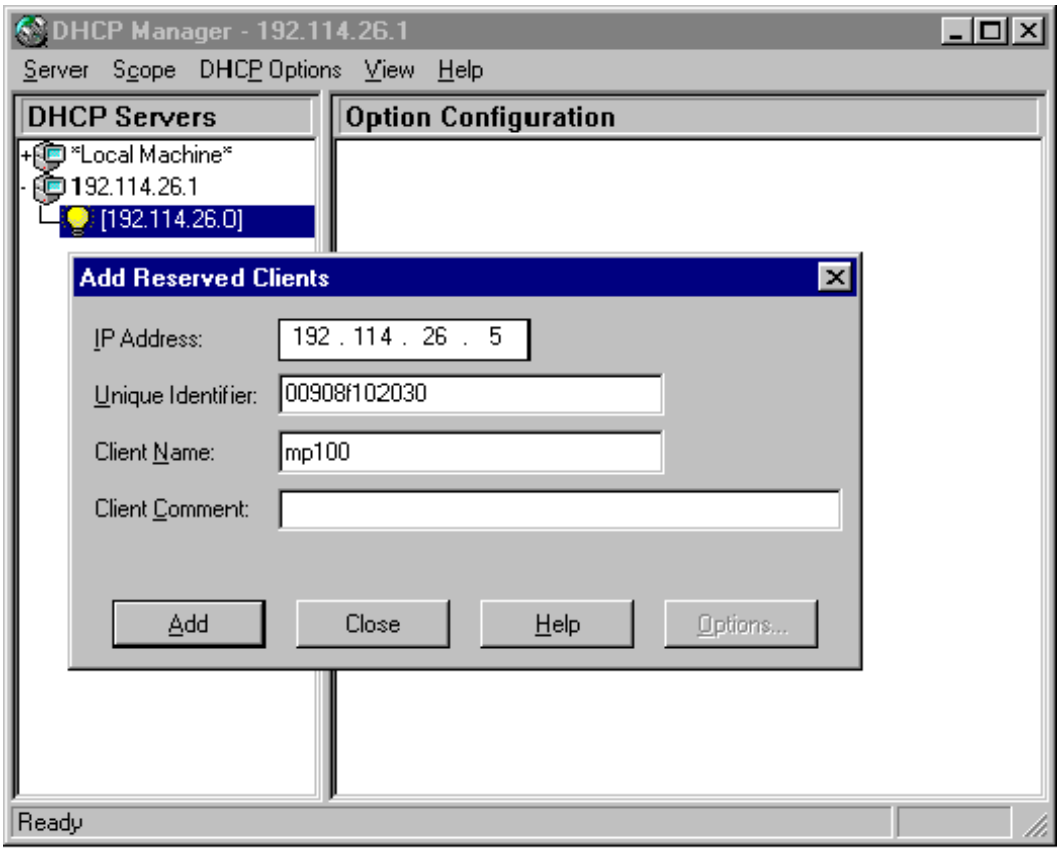
After starting the system, DHCP Manager displays the following screen:

Figure B-1: Scope Menu



1. Select “Add Reservations” from the “Scope” menu; the ‘Add Reserved Clients’ screen opens.

Figure B-2: Add Reserved Clients



2. Enter the IP address you want to provide to the AudioCodes Media Gateway.

3. The IP address reservation should be inside your DHCP Scope.

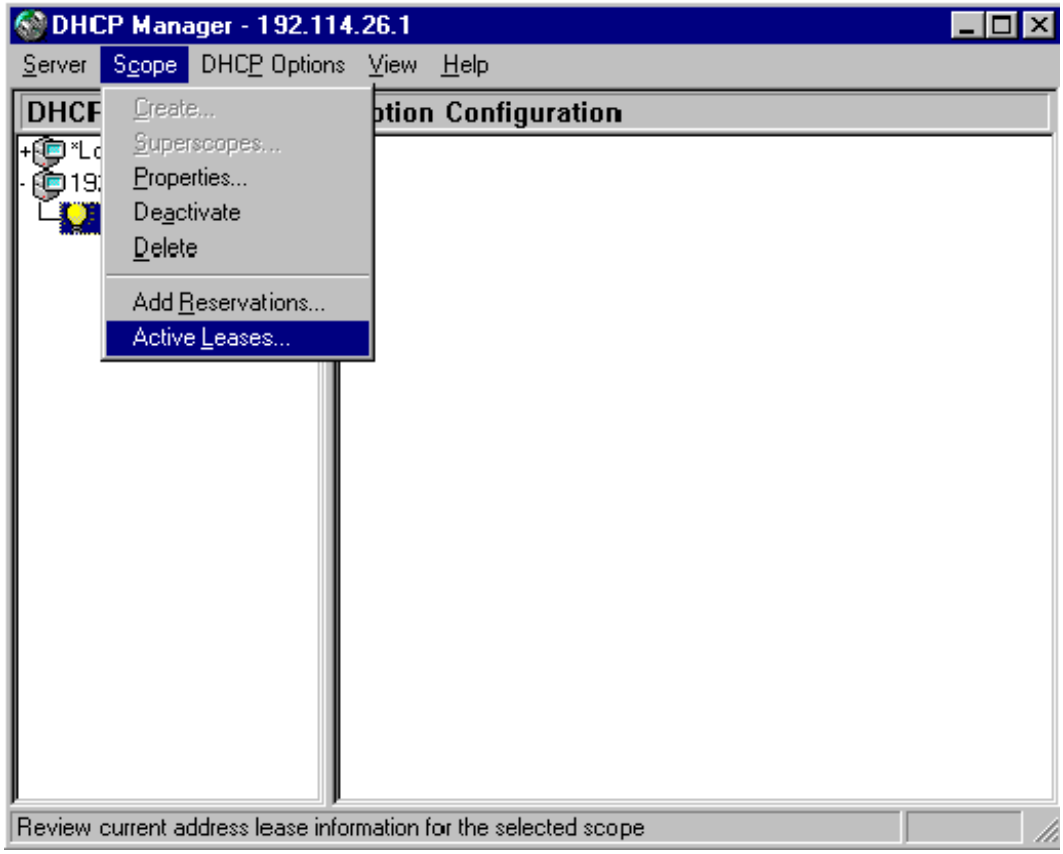
4. Enter the hardware MAC address, 12 digits, from your AudioCodes Gateway (provided in AudioCodes' Media Gateway MP-1xx/MP-200 and TP board/module product documentation.

5. Enter Client name; it can be any free text.

6. Select "Add" and then "Close" button.

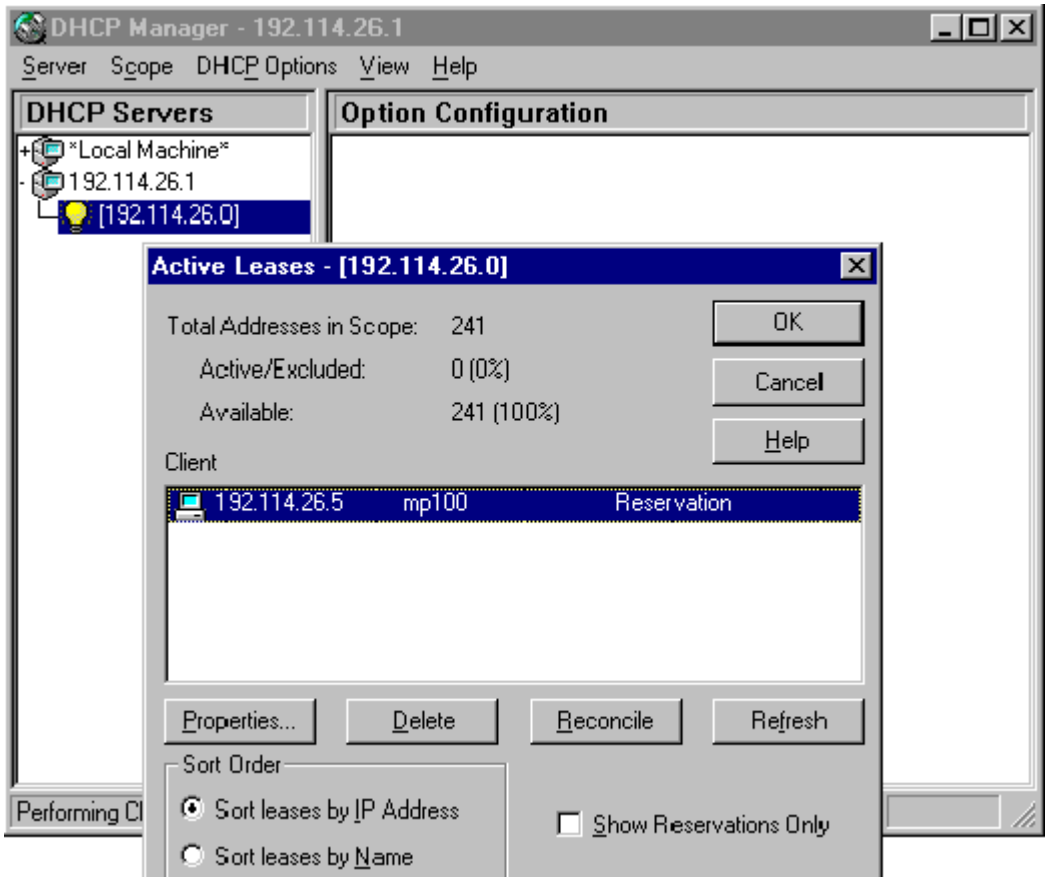
7. From Scope menu in the main screen, select “Active Leases”.

Figure B-3: Active Leases Select Screen



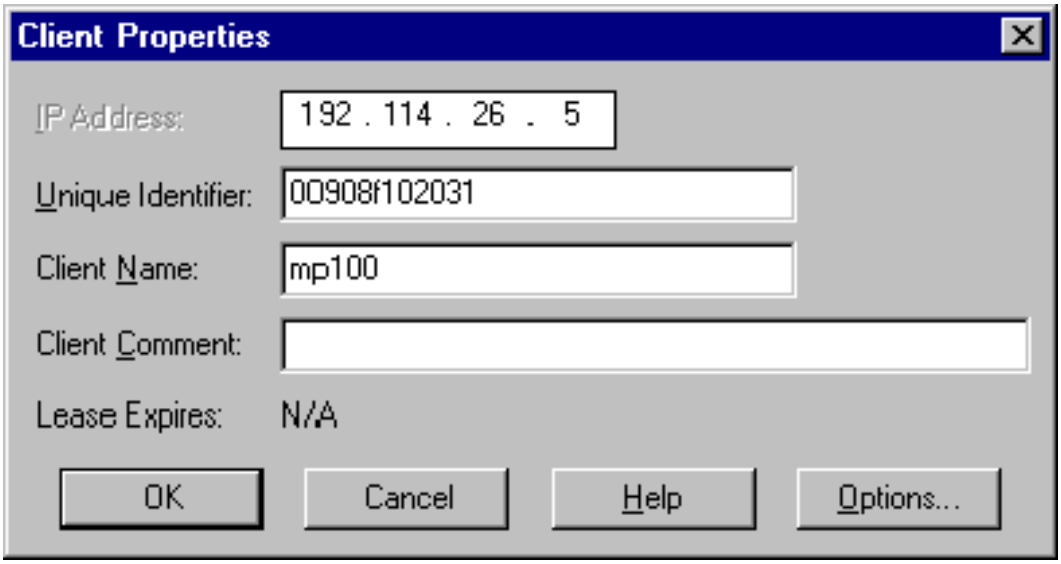
The Active Leases screen displays the following selection box:

Figure B-4: Active Leases Selection Box



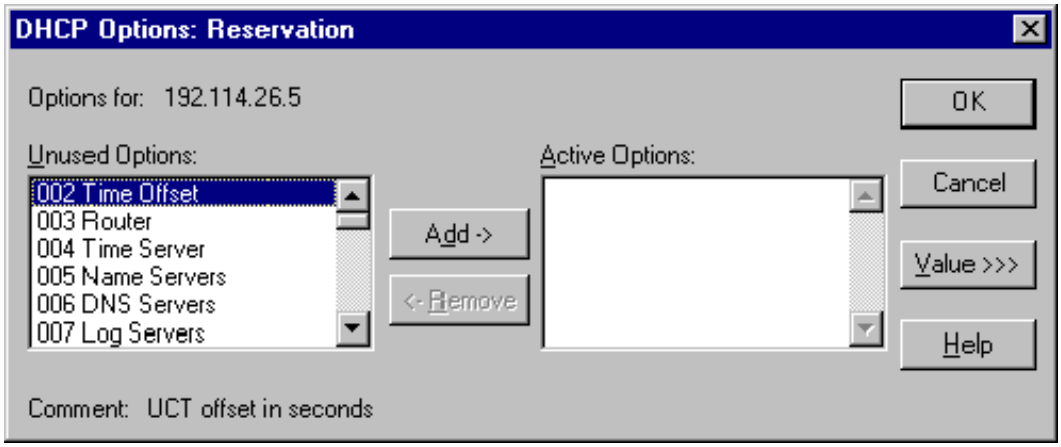
8. Press the Properties button.

Figure B-5: Client Properties Screen



9. Press the “Options” button to display the ‘DHCP Options: Reservation’ screen.

Figure B-6: DHCP Options: Reservation Screen.



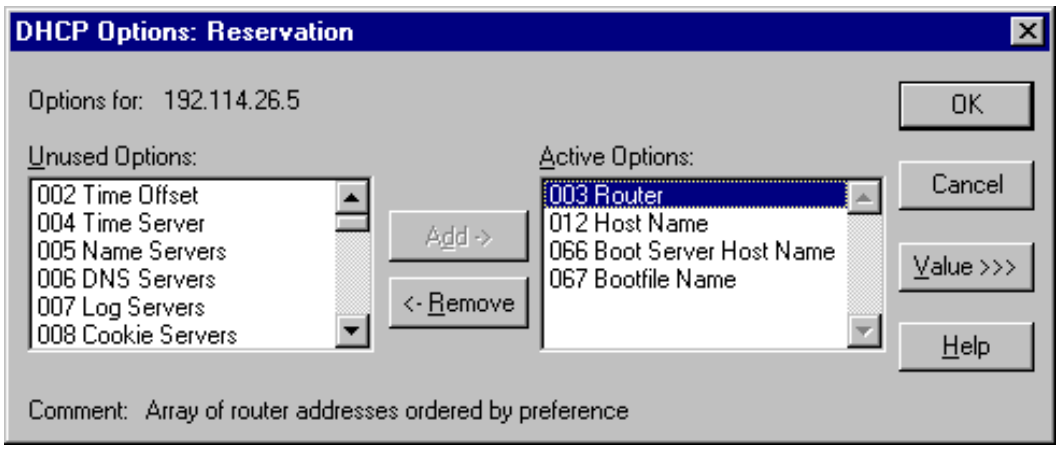
10. Add the following extension fields from the list in the left pane of the 'DHCP Options: Reservation' screen:

003 Router	(Default router/Gateway)
012 Host Name	(BootP client name such as **MP-200** or other customer selected name)
066 Boot Server Host Name	(Domain name or IP address of the TFTP server)
067 Bootfile Name	(Such as "ram.cmp")

Note: The file **ram.hex** is a loadable software image file. The file **ram.cmp** is a compressed version of the ram.hex, enabling faster download and reduced file size. The ram.cmp is the only version that can be burned to the flash memory.

The selected extension fields are then displayed in the right pane of the 'DHCP Options: Reservation' screen under 'Active Options', as shown below:

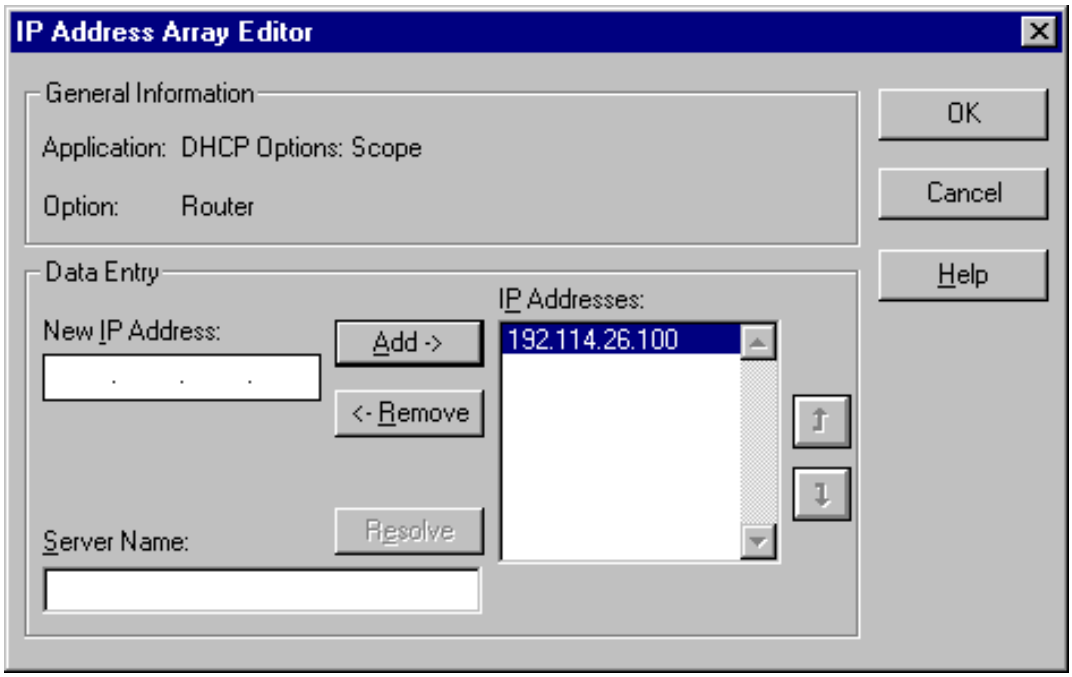
Figure B-7: Active Options



11. Select the "003 Router" field, as shown above, and click on the "Value" button.

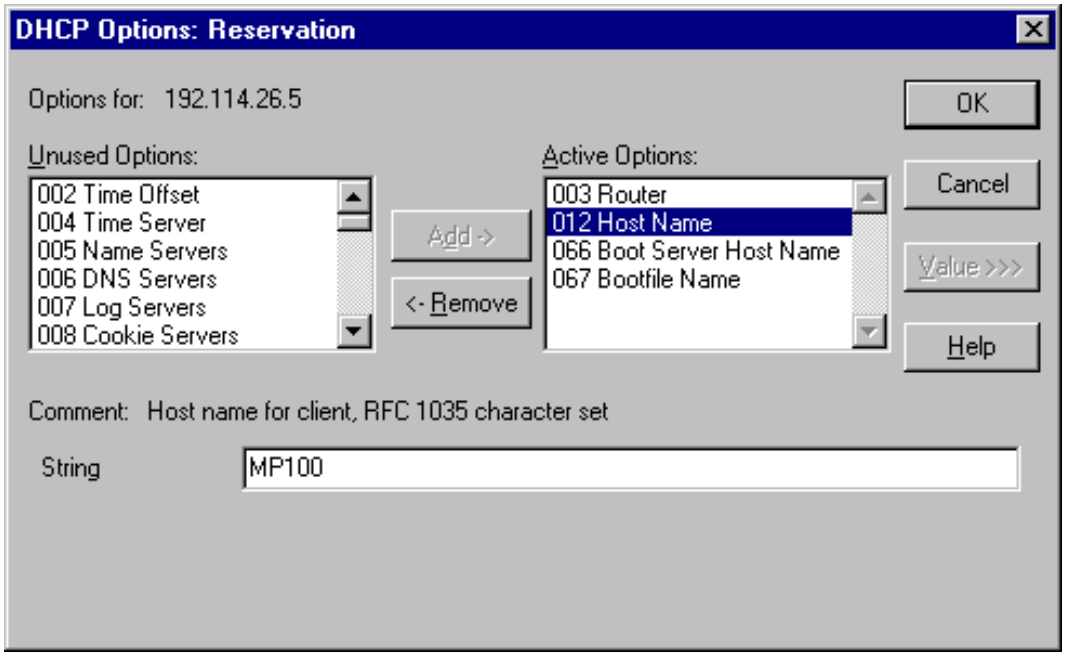
12. Edit the IP Address Array in the 'IP Address Array Editor' screen.

Figure B-8: IP Address Array Editor Screen



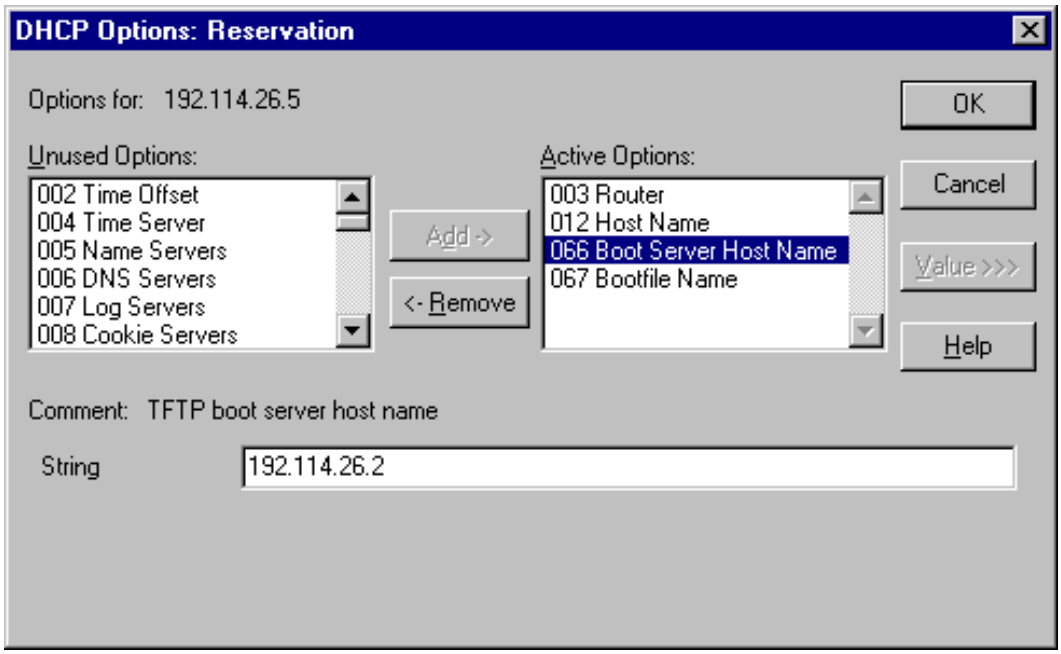
13. Click “OK”, and set the value for the Host name, which can be any character string.

Figure B-9: Host Name Screen



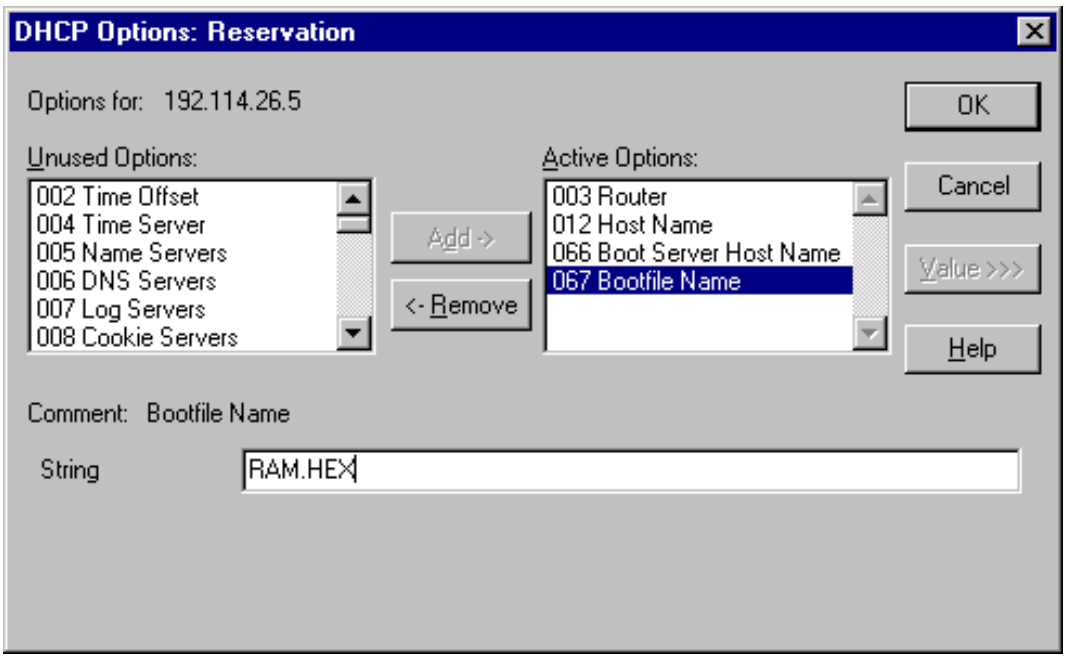
14. Click “OK”, and set the value, the IP address for the Boot Server Host Name. (That is, the server where the TFTP server is installed.)

Figure B-10: Boot Server Host Name Screen



15. Click “OK” and set the value, the Boot file name for the TFTP downloaded file.

Figure B-11: Bootfile Name Screen



16. Click “OK”, and exit from the DHCP server installation.

Appendix C- BootP Server Installation & Configuration

This section describes the Installation & Configuration of the BootP Server.

C.1 BootP Server Installation

The BootP Server95 can be downloaded from www.weird-solutions.com; it can be installed on Windows™ 95/98 or Windows™ NT.

C.2 BootP Server Configuration

➢ To configure the BootP server take the next 18 steps:

1. Start the BootP Server 95 application from the Programs menu on a PC running Windows™ 95/98 or Windows™ NT. The following screen appears:

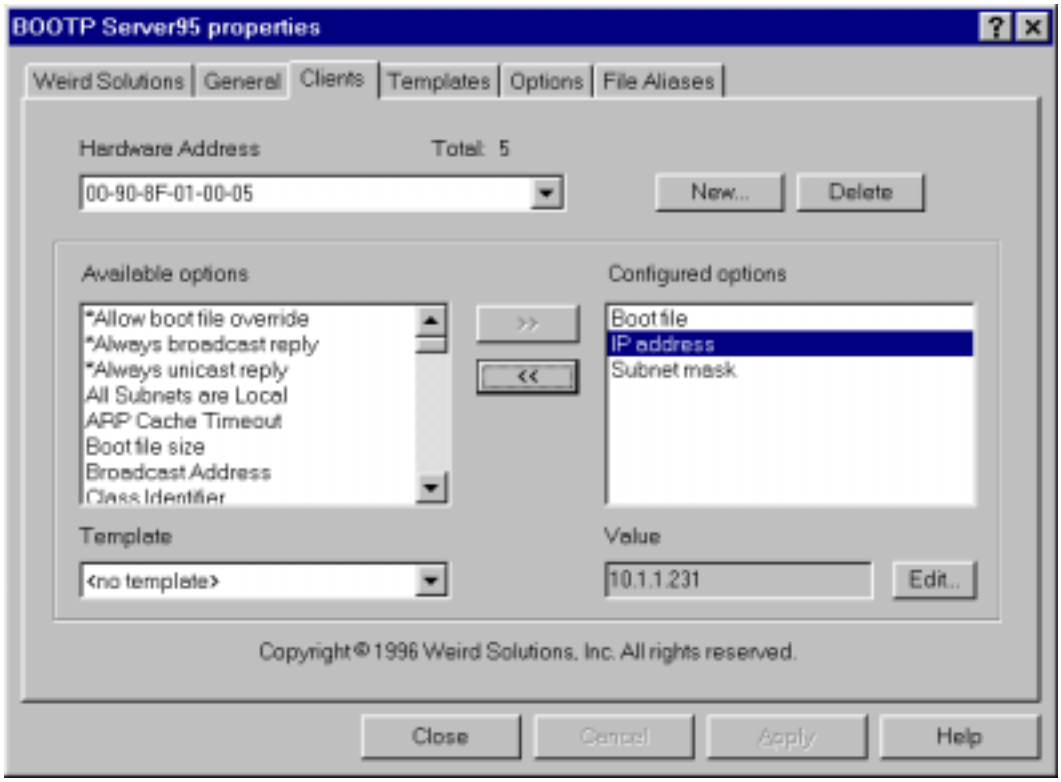
Figure C-1: BOOTP Server95

2. In the “Service” menu, select “Properties” (as shown in Figure C-1 above); the ‘BOOTP Server95 properties’ screen opens, as shown in Figure C-2 below.

Figure C-2: Properties Screen

3. In the ‘BOOTP Server95 properties’ screen, click tab “Clients”; the Available options’ screen opens, as shown in Figure C-3 below.

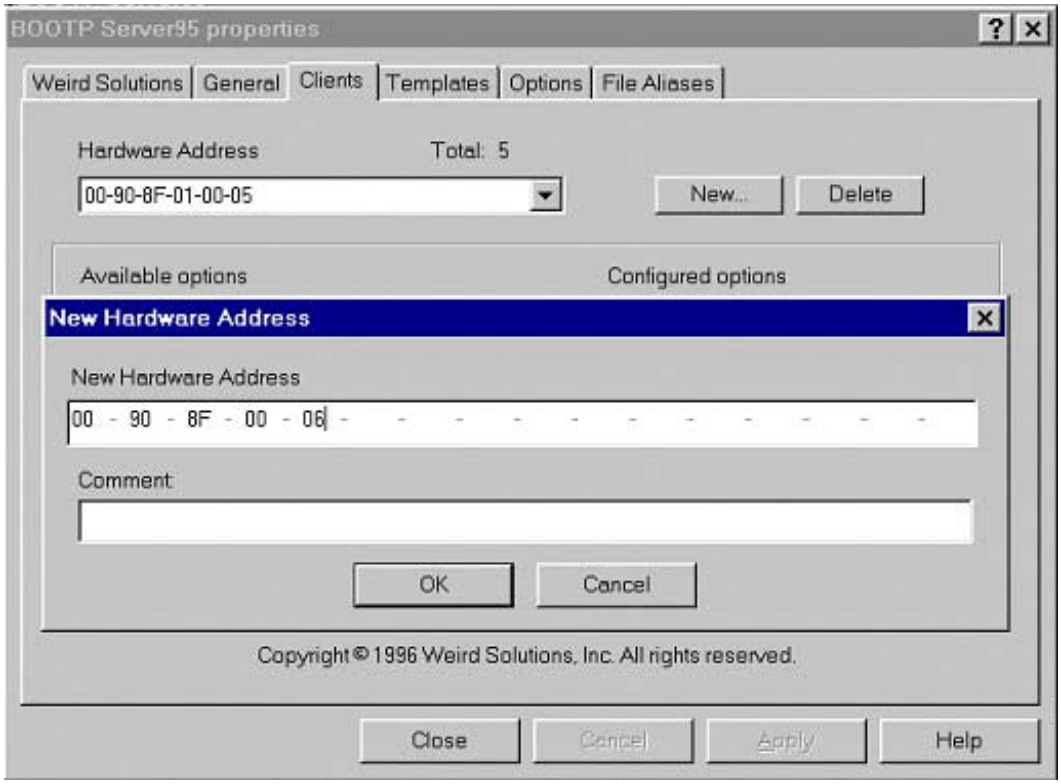
Figure C-3: Available Options Screen



4. Click on the button “New” in order to define a new Hardware Address (MAC address), or select an already defined Hardware Address to modify previously configured options.

5. Enter the new Hardware Address of 12 digits in the 'New Hardware Address' screen (shown below in Figure C-4), and press OK. The Hardware Address can be found on a Label attached to the bottom of the AudioCodes unit. (It is also called the MAC address.)

Figure C-4: New Hardware Address



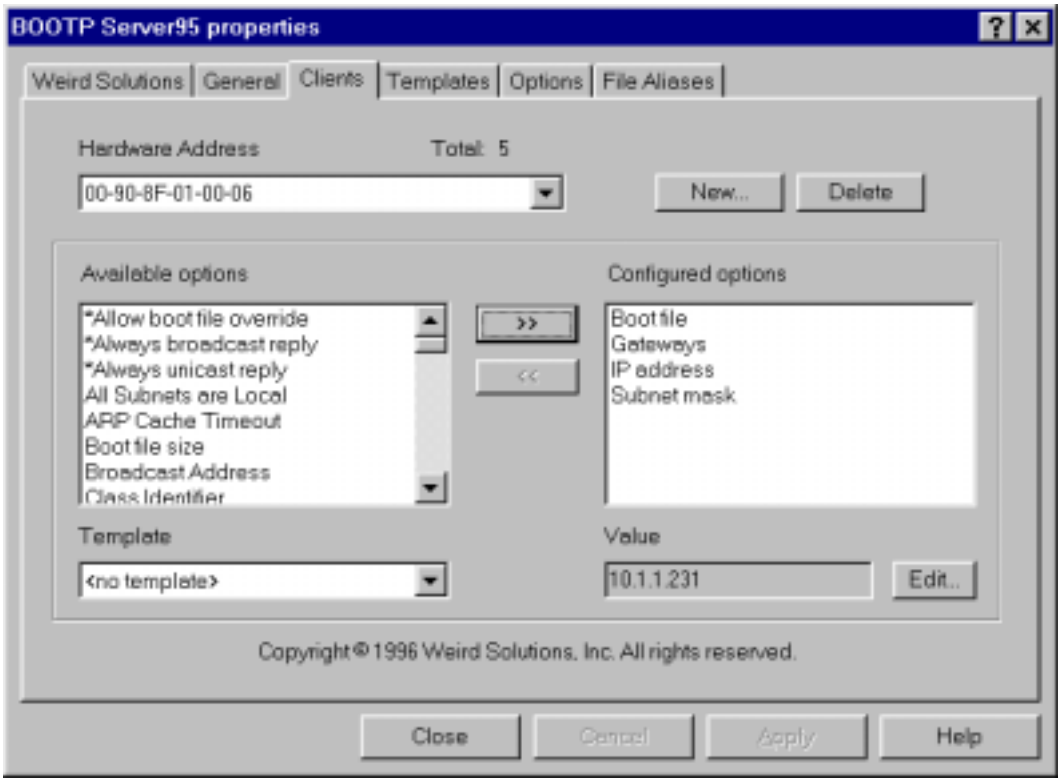
6. **Don't** fill the Comment field.

Refer to Section E-3 to change the given MAC address.

Now, in the 'Configured options' screen, you can see the new hardware address that you defined (see Figure C-5 below). In the left pane, you can see the "Available options".

7. Select the options by clicking on the arrows. The option you choose will be listed in the right pane under "Configured options". You can move back from any option by using the "back arrows".

Figure C-5: Configured Options Screen



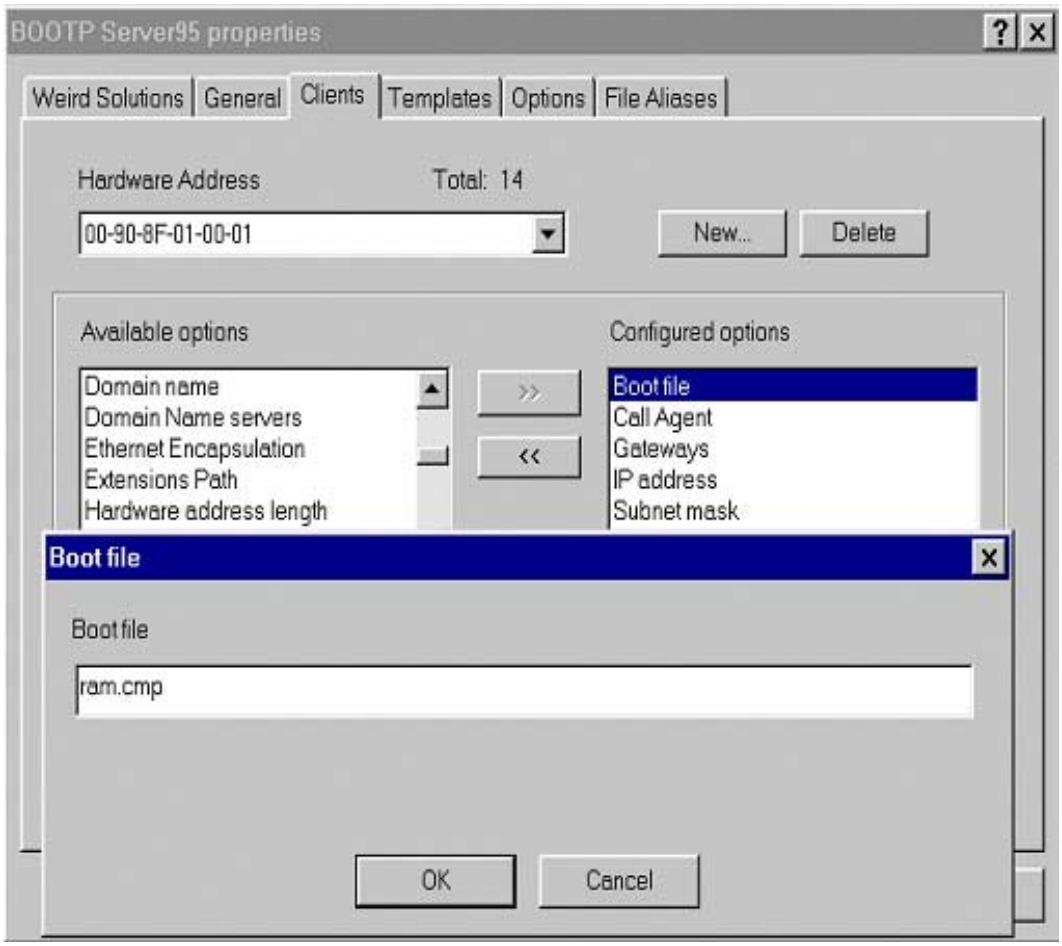
8. Select the following options listed under the 'Available options' pane, and move them to the 'Configured options' pane.
   - Boot file
   - Gateways
   - IP address
   - Subnet mask
   - Call Agent (see Section E-2 for setting the Call Agent option).

9. Set a value to each option.
   - Boot file - such as ram.cmp
   - Gateways - IP address of default Gateway, can be any address within the subnet
   - IP address - such as 10.1.1.33
   - Subnet Mask - such as 255.255.255.0 (see the restrictions in step 14.)
   - Call Agent - IP address of the Call Agent.

Note: The file **ram.hex** is a loadable software image file. The file **ram.cmp** is a compressed version of the ram.hex, enabling faster download and reduced file size. The ram.cmp is the only version that can be burned to the flash memory.

10. Select 'Boot file' from the list in the 'Configured options' pane in order to enter the Boot File name, and press the Edit button; the 'Boot File' screen opens, as shown in Figure C-6 below.

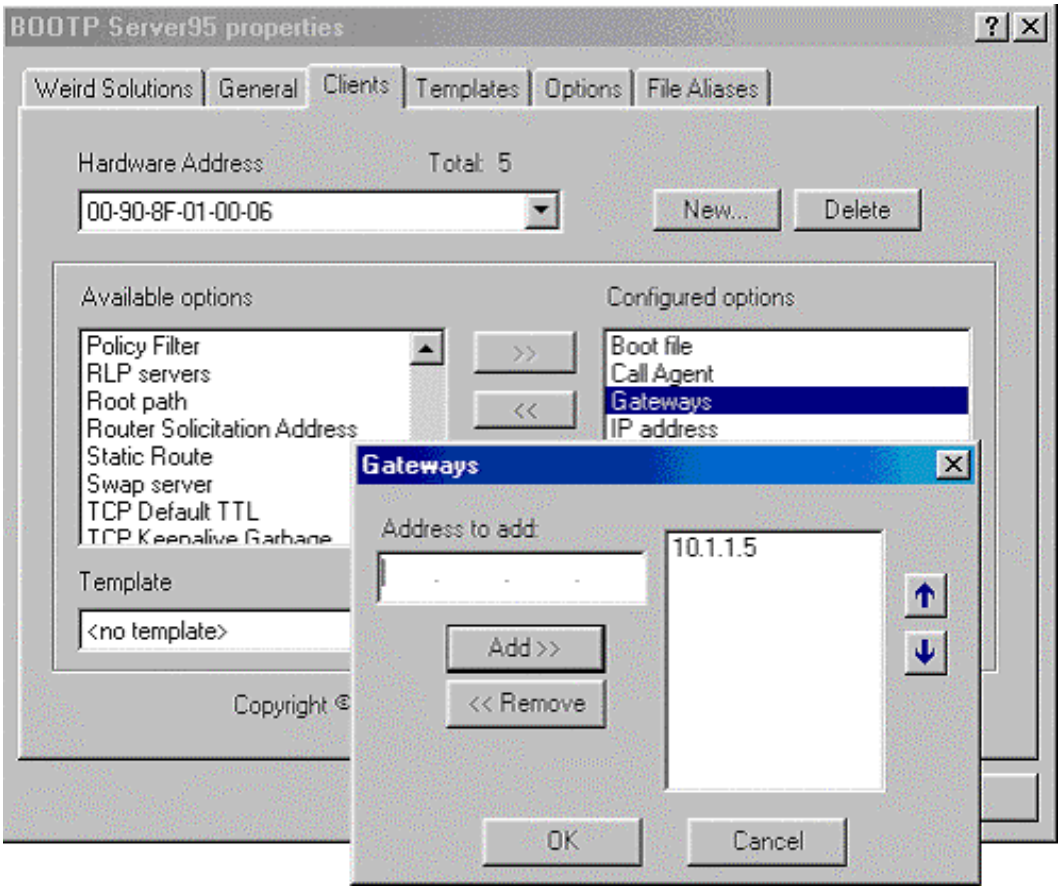
Figure C-6: Boot File Screen



11. Enter the boot file name and optional *ini* file name such as ram.cmp;mp200.ini. Click "OK".

12. Enter the Default Gateway IP address, by selecting Gateways and pressing the Edit button. The default Gateway address must be within your subnet.

Figure C-7: Default Gateway Screen

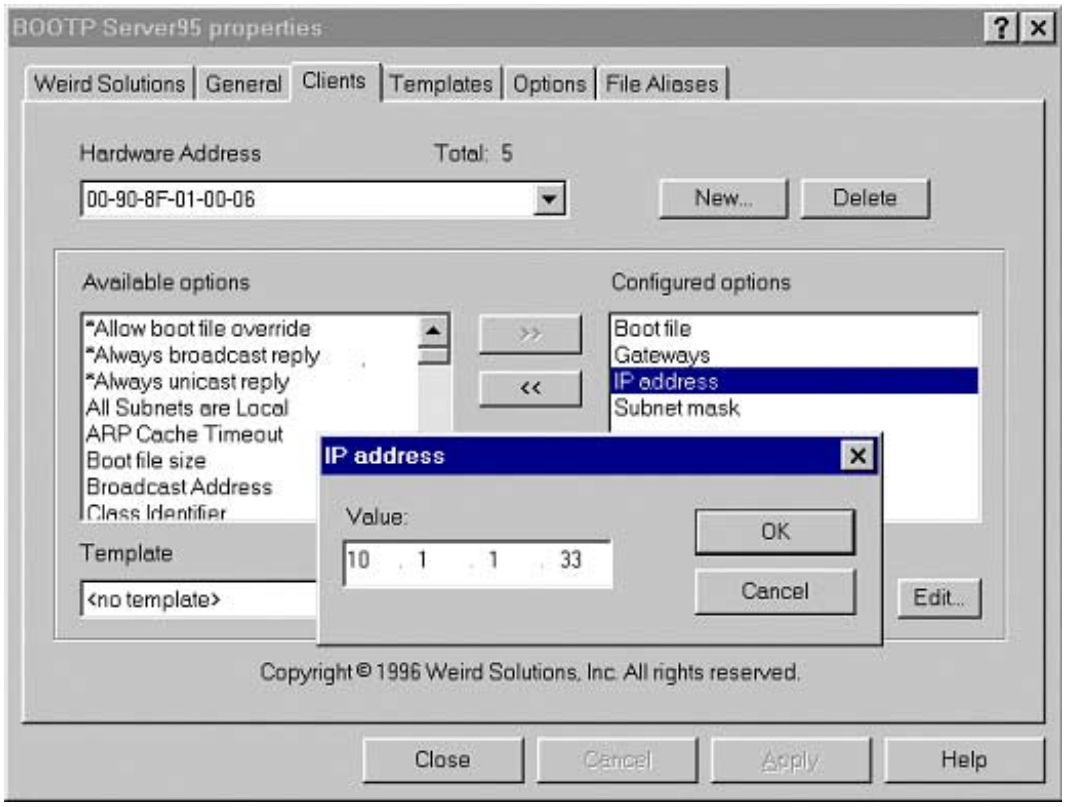


13. Enter the "Address to add", then press the "Add" button, and "OK".

Note: Only a single default Gateway can be used.

14. Enter the **MP unit** or **TP board/module** IP address.

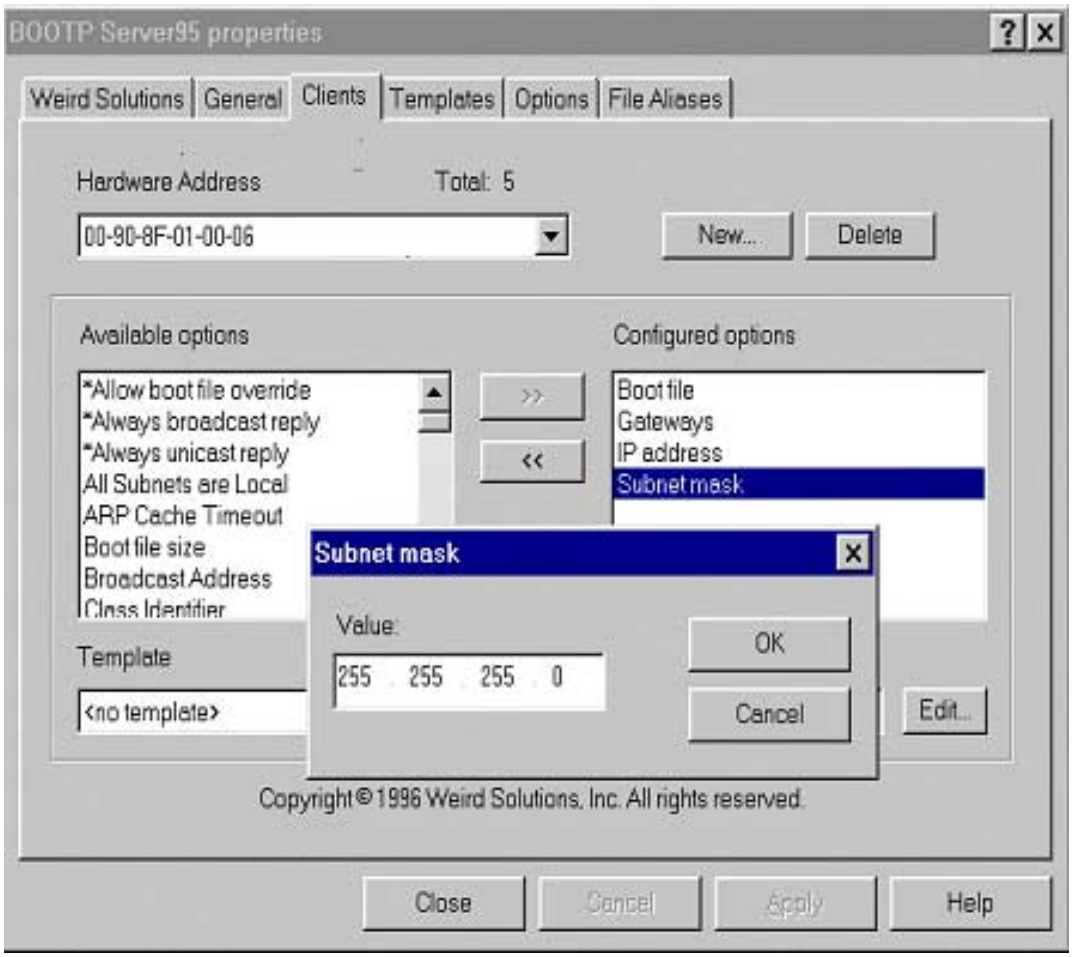
Figure C-8: MP Unit or TP Board/Module IP Address



15. Enter the Subnet Mask.

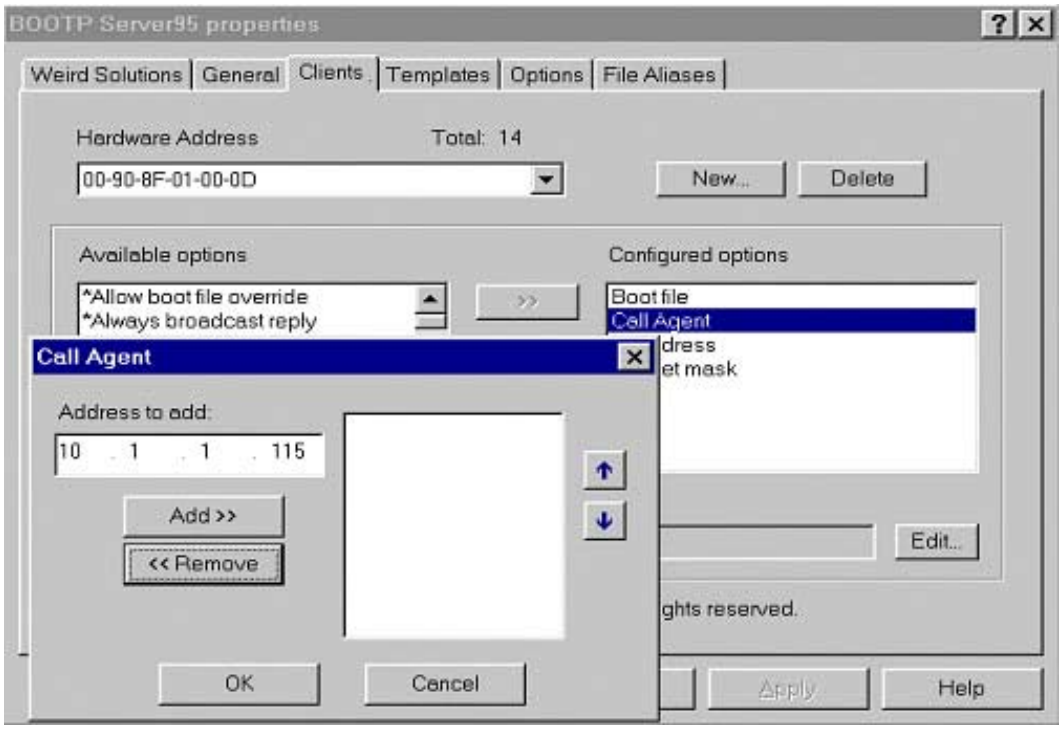
Set the Subnet Mask to a valid value in accordance with the IP address. (That is, class C IP addresses can only have a subnet mask starting with 255.255.255.X, while class B IP addresses can only have a subnet mask starting with 255.255.X.X, and class A IP addresses can only have a subnet mask starting with 255.X.X.X).

Figure C-9: Subnet Mask Screen



16. Enter the Call Agent IP Address, by selecting Call Agent and pressing the Edit button.

Figure C-10: Call Agent IP Screen

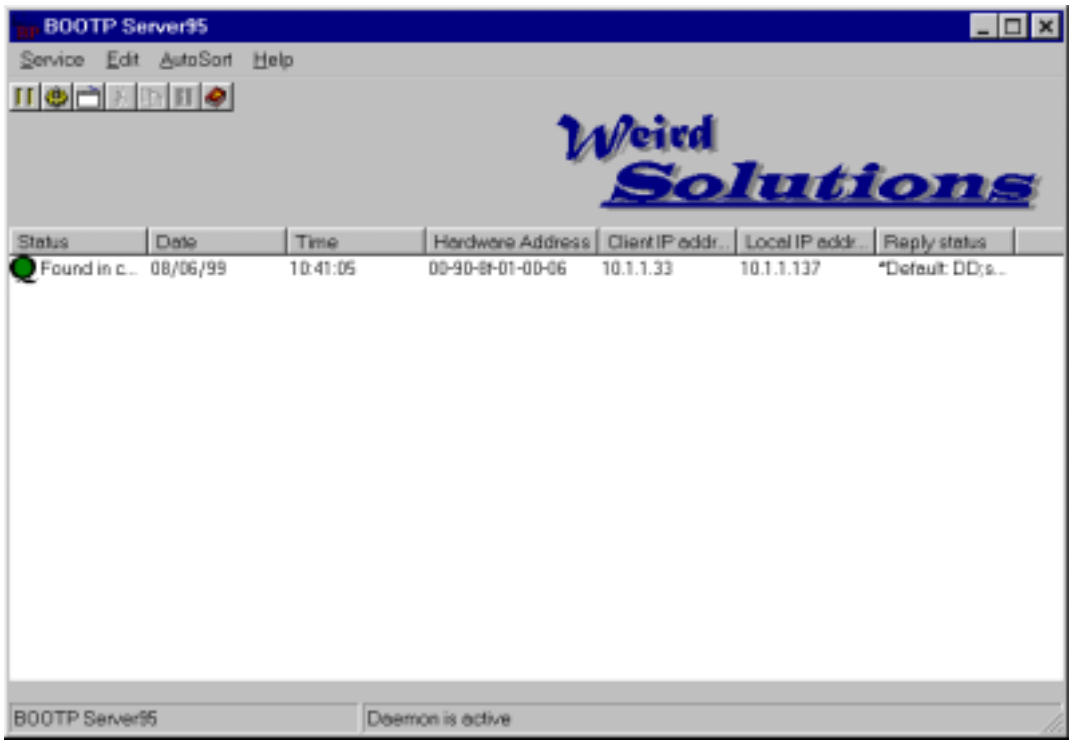


17. Enter the "Address to add", then press the "Add" button, and "OK".

18. Click "Close" to finish the BootP configuration.

After the MP-1xx/MP-200 or TP board/module is reset or powered up, the Status screen is displayed, as shown in Figure C-11 below:

Figure C-11: Status Screen


BOOTP Server95
Service Edit AutoSort Help
Weird Solutions
Status Date Time Hardware Address Client IP addr... Local IP addr... Reply status
Found in c... 08/06/99 10:41:05 00-90-8f-01-00-06 10.1.1.33 10.1.1.137 *Default: DD;s...
BOOTP Server95
Daemon is active

After the BootP process is successfully completed, the **MP unit or TP board/module** receives the name of the image file (ram.cmp), IP address, Subnet mask, default Gateway.

The next step is to download the image file (*.cmp file) from the TFTP server.

Note: The TFTP server and BootP servers should be installed on the same Host.

C.3 Setting Call Agent Address Option

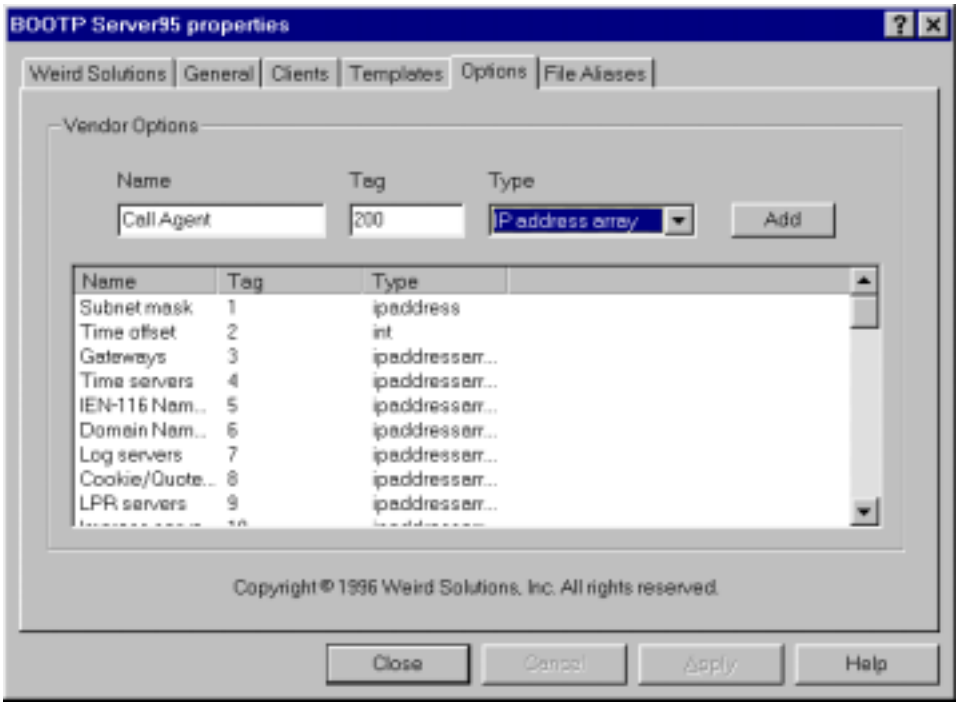
The Call Agent Address is required especially for MGCP RSIP (Restart in Progress) messages.

If the Call Agent address has not yet been defined in the list of the Available options, follow the next procedure.

➢ To run the Software Upgrade take the next 2 steps:

1. Click on tab 'Options' in the 'BOOTP Server95 properties' screen, and enter Name, Tag, and Type fields under the section 'Vendor options', as shown in Figure C-12 below. (Tag = 200, Type = IP address array).

Figure C-12: Call Agent (IP Address Array) Screen



2. Press the button 'Add', and 'OK'.

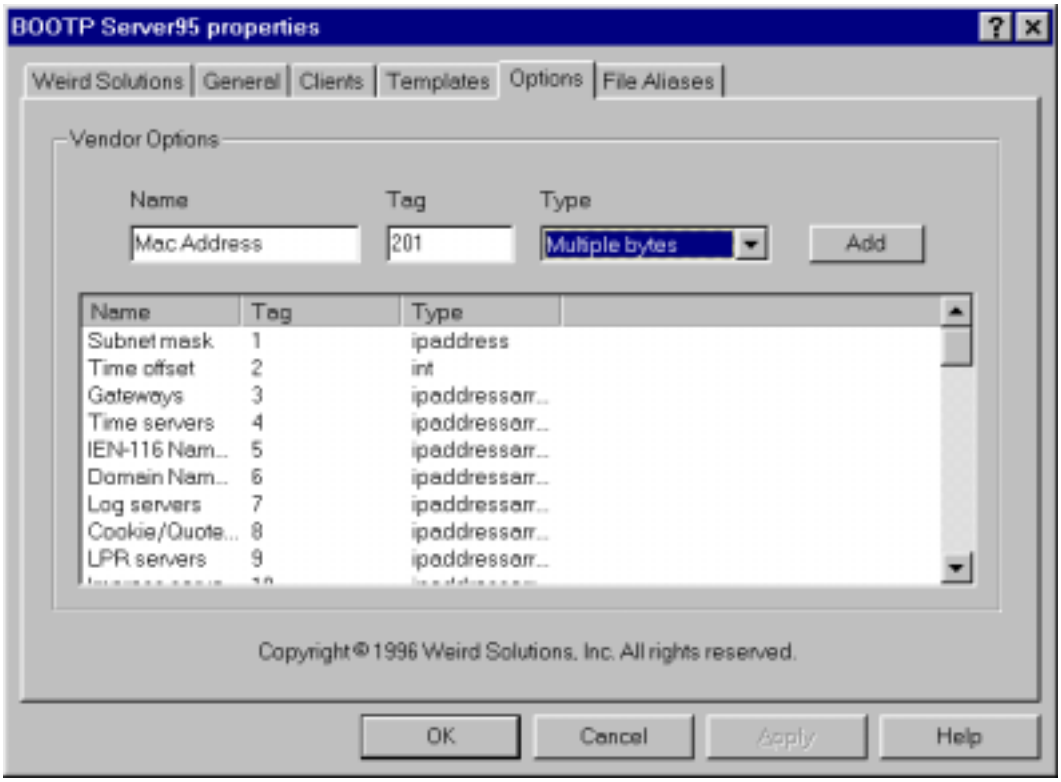
C.4 Changing MAC Hardware Address

If the MAC address has not yet been defined in the list of the 'Available options', follow the next procedure:

➢ **To change the MAC hardware address take the next 9 steps:**

1. Click tab 'Options' and under the section 'Vendor options', enter the Name, Tag, and Type fields, as shown in Figure C-13. (Tag = 201, Type = Multiple bytes).

Figure C-13: MAC (Multiple Bytes) Screen

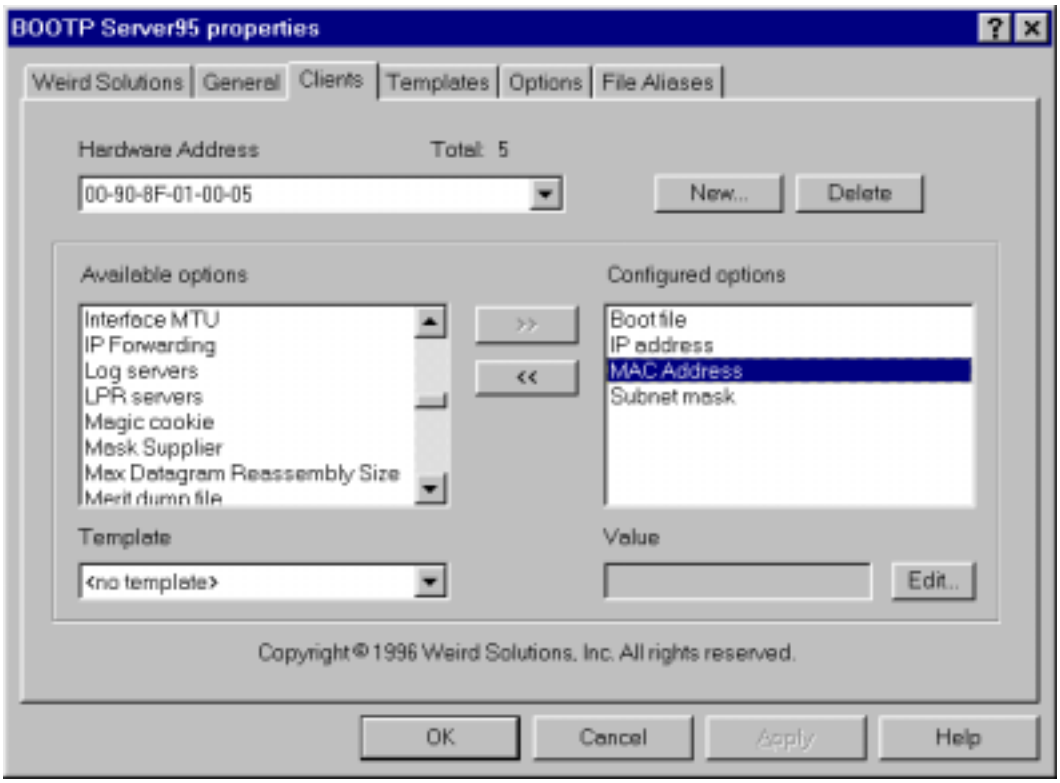


2. Press the button 'Add', and OK.

3. Set the value for a new MAC address.

4. Select MAC address listed in the 'Configured options' pane. In addition to MAC address, you need to configure and set a value for the IP address, but other parameters are not required. (For these procedures, the IP address can be any valid IP address.)

Figure C-14: MAC (Options) Screen

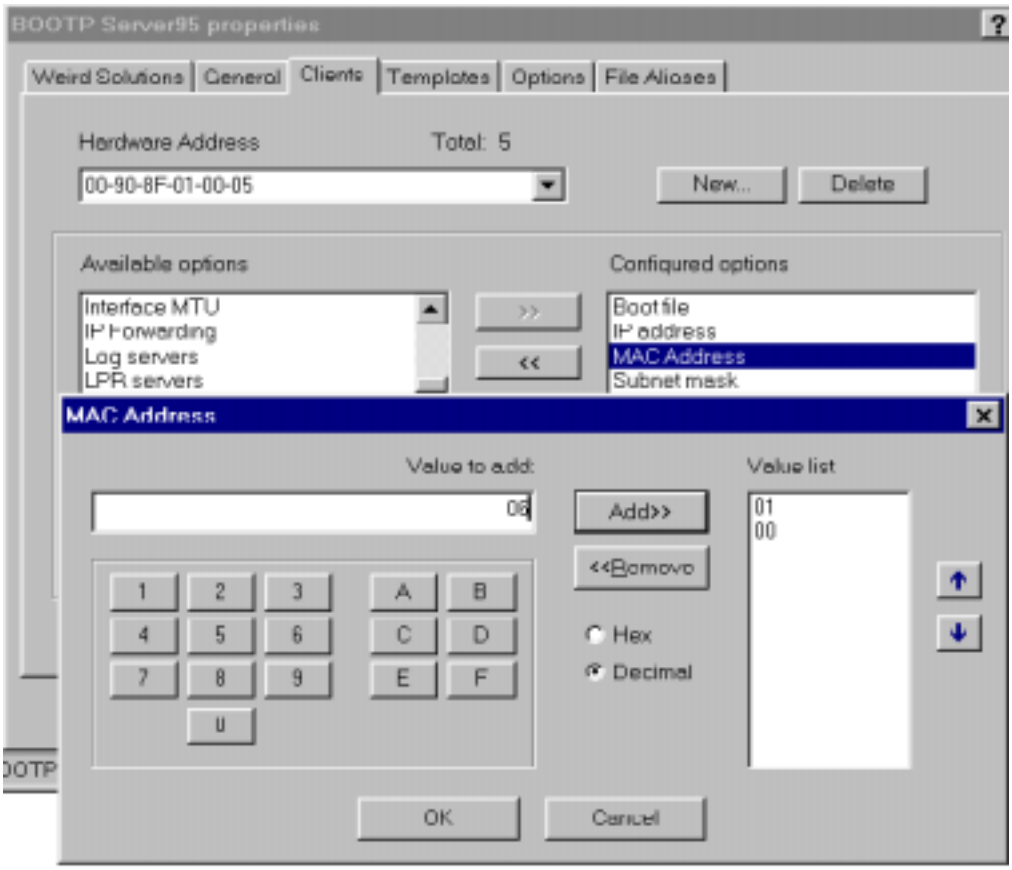


5. Click on the Edit button and in the 'MAC Address' screen (shown in Figure C-15 below), enter a value for the new MAC address. Enter six LSB digits such as 01 00 06. The MSB digits of the MAC address are fixed, and set in the software of the **AudioCodes MP unit** or **TP board/module** to **00 90 8F**.

6. After entering the LSB digits you will get the MAC address:

"**00 90 8F** 01 00 06".

Figure C-15: MAC Address Screen

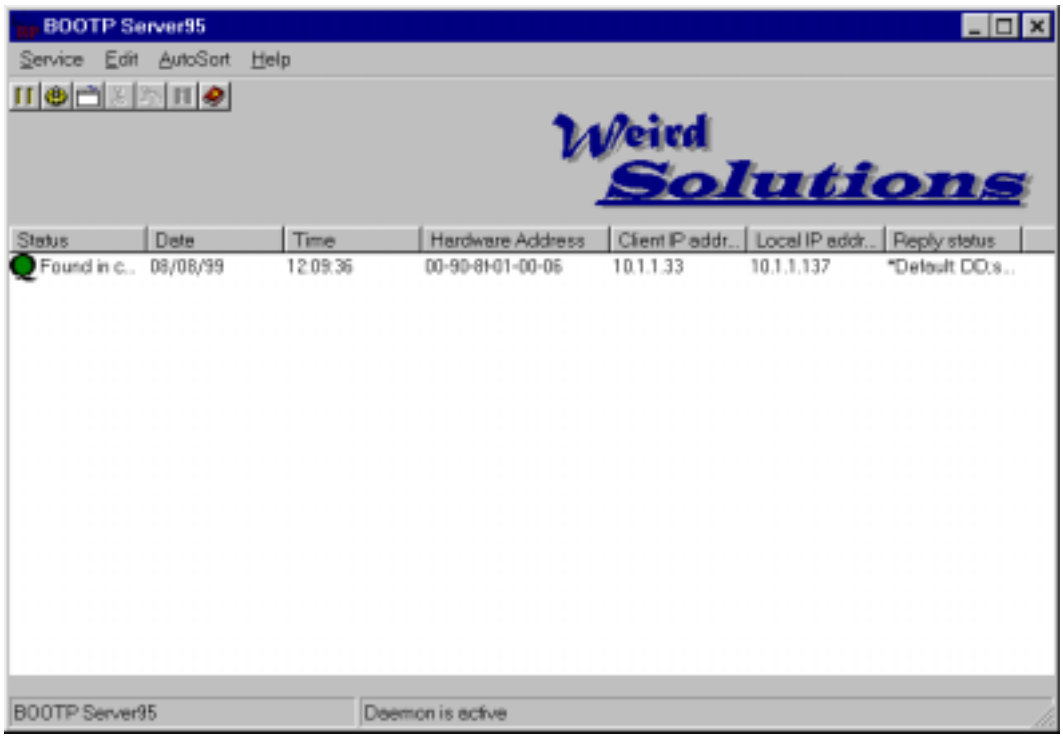


7. Press "OK" and then "Close".

8. To program the new MAC address in the **AudioCodes MP unit** or **TP board/module**, connect the **MP unit** or **TP board/module** to power or press its Reset button.

9. To validate the programming of new MAC address, press again on the Reset button and view the status message displayed on the RS-232 terminal screen (HyperTerminal), and on the 'BOOTP Server95' GUI screen.

Figure C-16: Status Validation Screen


BOOTP Server95
Service Edit AutoSort Help
Weird Solutions
Status Date Time Hardware Address Client IP addr.. Local IP addr.. Reply status
Found in c.. 08/08/99 12:09:36 00-90-8H01-00-06 10.1.1.33 10.1.1.137 *Default DO.s..
BOOTP Server95
Daemon is active

Appendix D– TFTP Server Installation & Configuration

This section describes the Installation & Configuration of the TFTP Server.

D.1 Install TFTP Server

The TFTP server ("TFTP Turbo 98") can be downloaded from www.weird-solutions.com; it can be installed on Windows™ 95/98 or Windows™ NT.

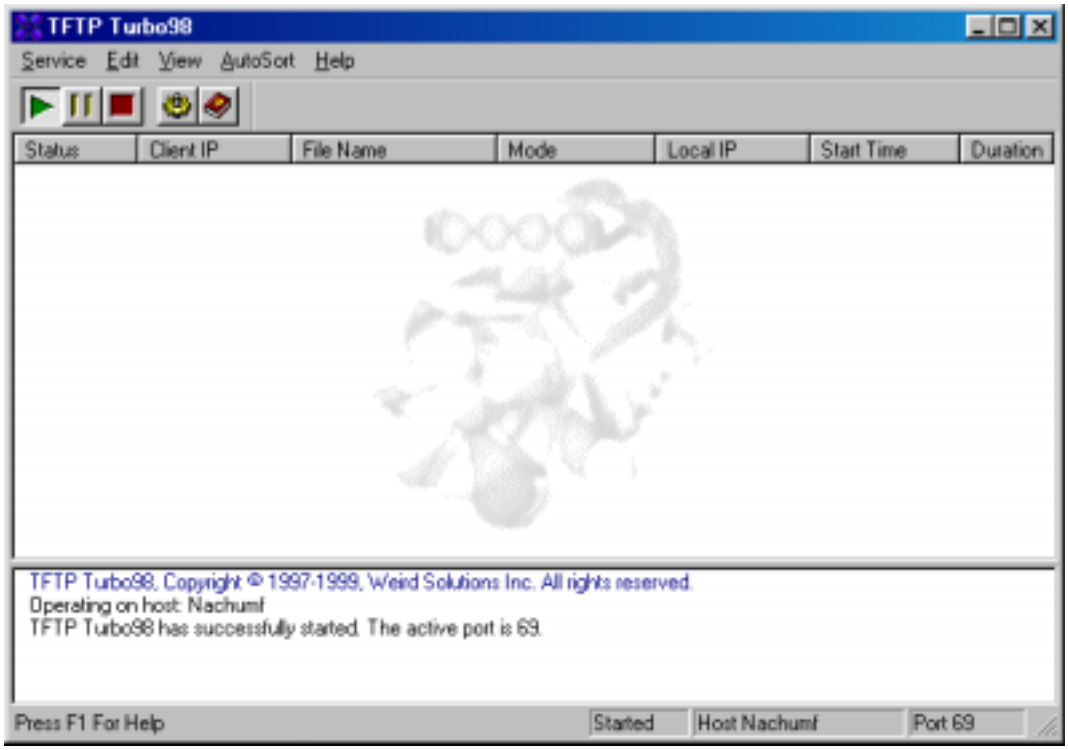
Note: The TFTP server and BootP servers should be installed on the same Host.

D.2 Configure TFTP Server

➢ To install and configure the TFTP server take the next 6 steps:

1. After entering the TFTP server application, the initial screen, TFTP Turbo98 is displayed, as shown in Figure D-1 below:

Figure D-1: TFTP Initial Screen

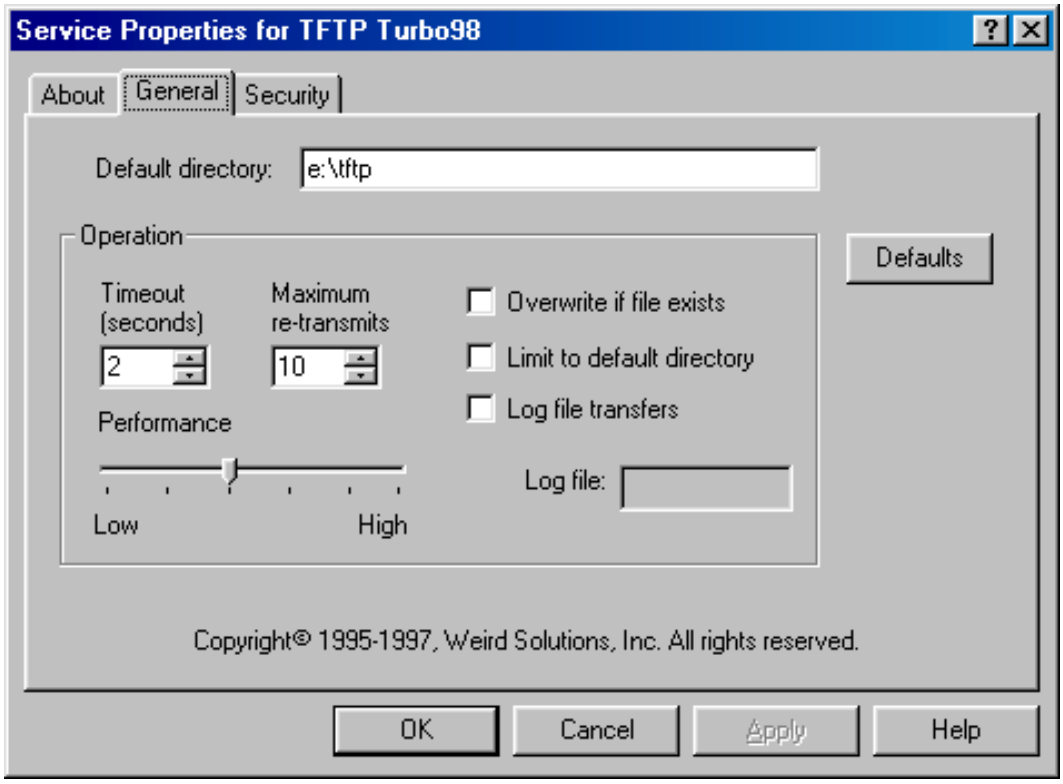


2. In the TFTP Turbo98 initial screen, open the “Service” menu and choose “Properties”; the ‘Service Properties for TFTP Turbo98’ screen is displayed, as shown in Figure D-2 below:

Figure D-2: Service Properties Screen

3. Press tab ‘General’; the TFTP General screen, shown in Figure D-3 on the next page, is displayed.

Figure D-3: TFTP General Screen



4. In the “Default directory’ field, enter the default directory where the **ram.cmp** file is located. Don't forget to copy the file to that directory before starting to use the TFTP server.

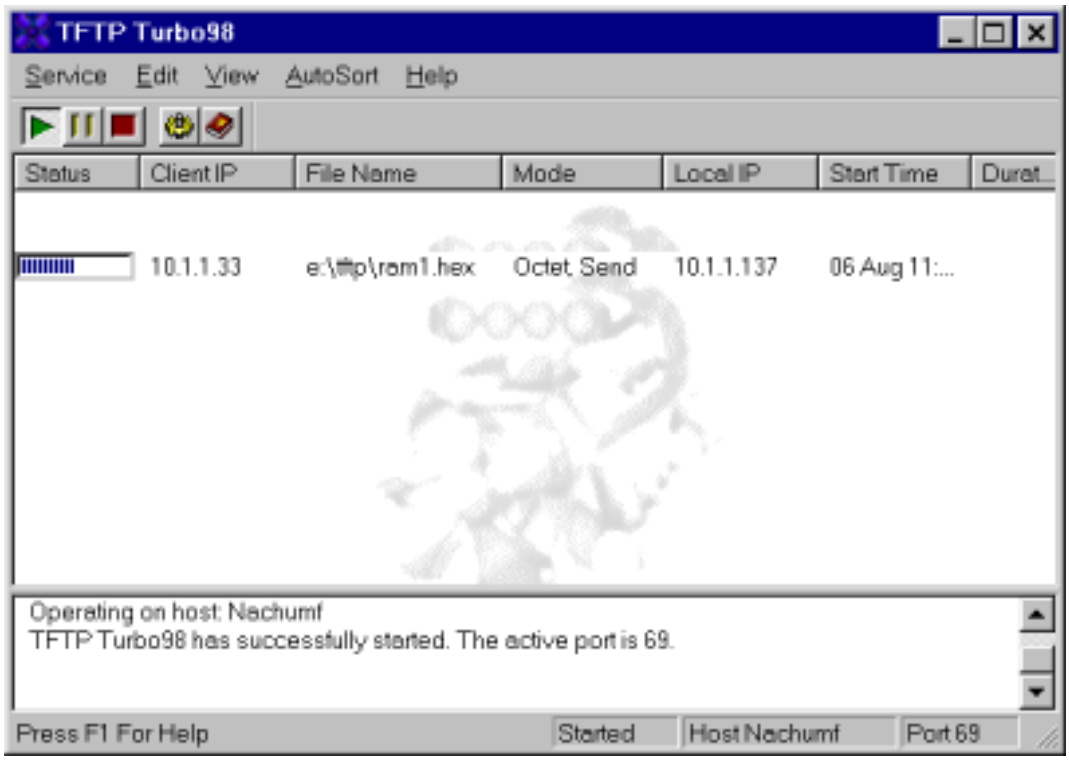
5. Set ‘Timeout’ to 2 seconds; set ‘Maximum re-transmits’ to 10.

6. Select "Close" button to finish the TFTP server installation.

During the starting procedure of the **AudioCodes MP unit** or **TP board/module**, its TFTP client requests the download of the ram1.cmp and an optional mp200.ini file from the TFTP server.

Finally, you'll view the status screen shown in Figure D-4 below.

Figure D-4: TFTP Final Status Screen


TFTP Turbo98
Service Edit View AutoSort Help
Status Client IP File Name Mode Local IP Start Time Durat...
10.1.1.33 e:\tftp\rom1.hex Octet Send 10.1.1.137 06 Aug 11:...
Operating on host: Nachumf
TFTP Turbo98 has successfully started. The active port is 69.
Press F1 For Help
Started
Host Nachumf
Port 69

Reader's Notes

Appendix E - Call Manager Installation & Configuration

Version 2.6

This section describes the Installation & Operation of the Call Manager Version 2.6.

E.1 Software Installation Procedure

Call Manager can be installed on Windows™ NT, Windows 2000 or Windows™ 98.

To install Call Manager, run the Setup.exe file from the package supplied. The package can be on five diskettes of 1.44 MB each, or a CD-ROM package containing 10 files (total size ~ 6 MB).

During the installation procedure, the User is prompted to answer typical installation questions (destination directory, folder name, and so on).

After the installation procedure, the destination directory will contain the program's executable file CallAgent.exe, the Log file CallManager.log and the directory "data" containing the database's definitions.

E.1.1 Installing Call Manager as a Service

The Call Manager can be installed as a Service on Windows NT 4.0 or Windows 2000.

➢ To install Call Manager as a Service take the next 3 steps:

1. Find utilities XYNTService.ini and XYNTServiceInstall.bat in the home directory of Call Manager.
2. Edit the XYNTService.ini file and set the 'CommandLine' parameter to be a full path to the CallManager.exe file. For example: CommandLine=C:\Program Files\AudioCodes\CallManager\CallAgent.exe

Note:	Do NOT change 'ServiceName' parameter, because its value is hard coded.

3. Run utility XYNTServiceInstall.bat; the installation is finished; on the computer's next reboot, Call Manager will start automatically.

➢ To uninstall the Call Manager from Service, take the next step:

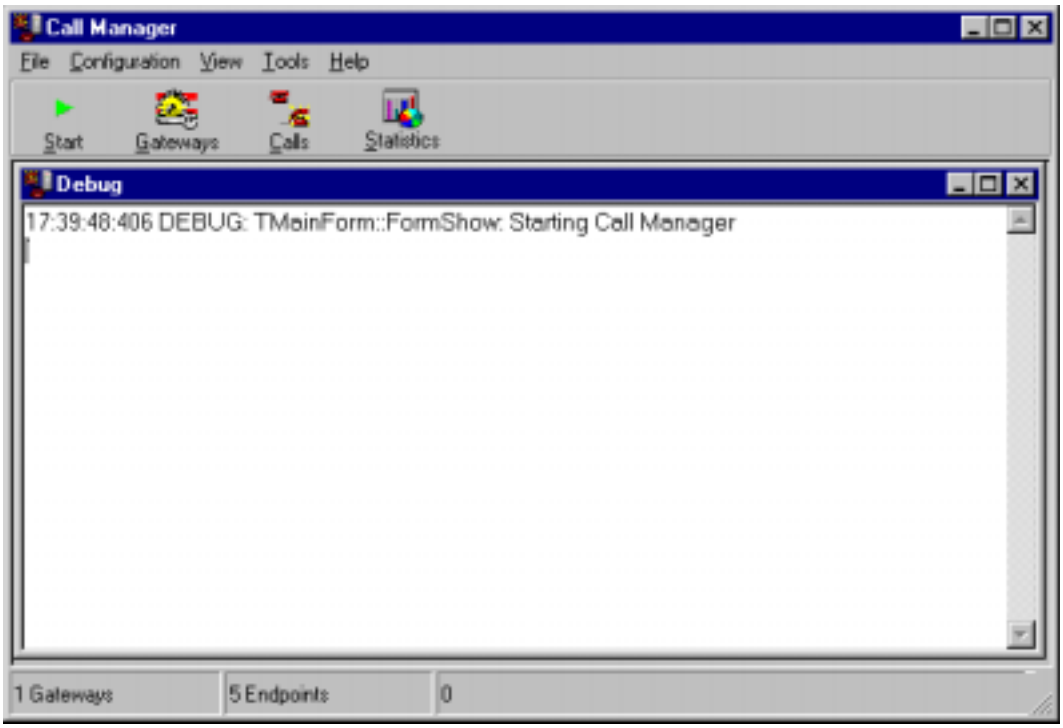
1. Run the utility XYNTServiceUninstall.bat from the Call Manager's home directory.

Note: While working with Call Manager as Service, avoid running the program from the Start menu. It may cause ambiguity and the program's failure.

E.2 Call Manager Application

Start Call Manager from the Programs menu; the following window appears:

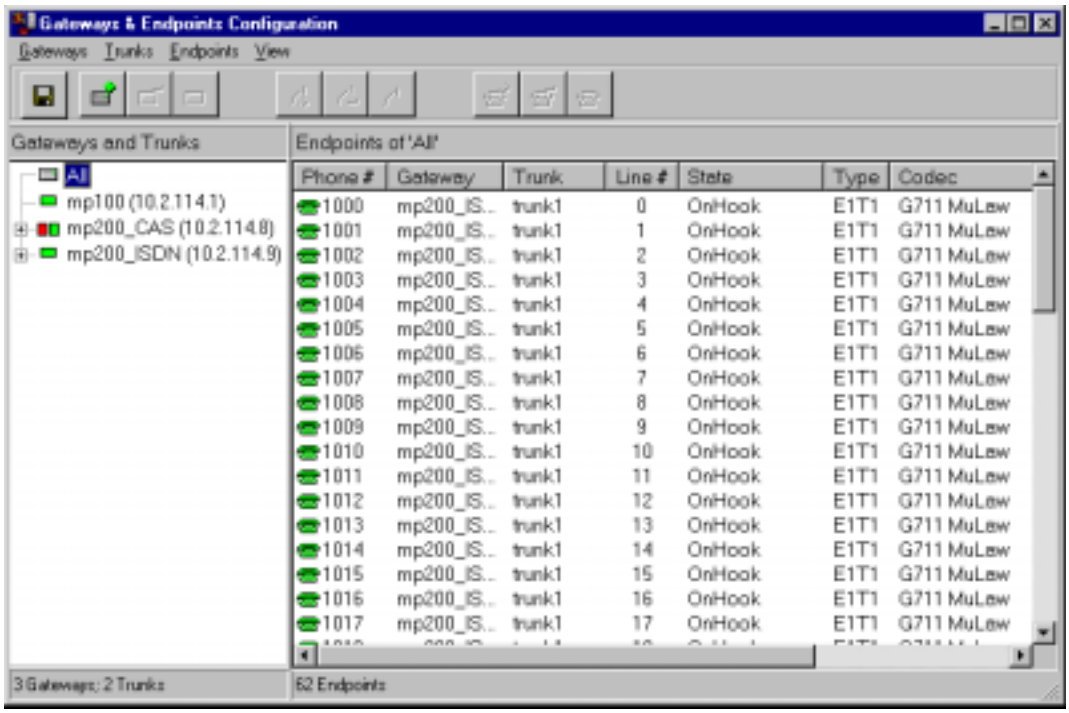
Figure E-1: Call Manager Main Window



E.2.1 Gateways and Endpoints Configuration

In Call Manager's main window, select "Configuration" and then "Gateways & Endpoints"; the Gateways & Endpoints Configuration window opens:

Figure E-2: Gateways & Endpoints Configuration


Gateways & Endpoints Configuration
Gateways Trunks Endpoints View
Gateways and Trunks
All
mp100 (10.2.114.1)
mp200_CAS (10.2.114.8)
mp200_ISDN (10.2.114.9)
Endpoints of 'All'
Phone # Gateway Trunk Line # State Type Codec
1000 mp200_IS.. trunk1 0 OnHook E1T1 G711 MuLaw
1001 mp200_IS.. trunk1 1 OnHook E1T1 G711 MuLaw
1002 mp200_IS.. trunk1 2 OnHook E1T1 G711 MuLaw
1003 mp200_IS.. trunk1 3 OnHook E1T1 G711 MuLaw
1004 mp200_IS.. trunk1 4 OnHook E1T1 G711 MuLaw
1005 mp200_IS.. trunk1 5 OnHook E1T1 G711 MuLaw
1006 mp200_IS.. trunk1 6 OnHook E1T1 G711 MuLaw
1007 mp200_IS.. trunk1 7 OnHook E1T1 G711 MuLaw
1008 mp200_IS.. trunk1 8 OnHook E1T1 G711 MuLaw
1009 mp200_IS.. trunk1 9 OnHook E1T1 G711 MuLaw
1010 mp200_IS.. trunk1 10 OnHook E1T1 G711 MuLaw
1011 mp200_IS.. trunk1 11 OnHook E1T1 G711 MuLaw
1012 mp200_IS.. trunk1 12 OnHook E1T1 G711 MuLaw
1013 mp200_IS.. trunk1 13 OnHook E1T1 G711 MuLaw
1014 mp200_IS.. trunk1 14 OnHook E1T1 G711 MuLaw
1015 mp200_IS.. trunk1 15 OnHook E1T1 G711 MuLaw
1016 mp200_IS.. trunk1 16 OnHook E1T1 G711 MuLaw
1017 mp200_IS.. trunk1 17 OnHook E1T1 G711 MuLaw
3 Gateways; 2 Trunks
62 Endpoints

The two-pane window "Gateways & Endpoints Configuration", shown above, enables the configuration of *Gateways/Trunks* and *Endpoints* to be controlled by Call Manager.

The left pane represents all Gateways and Trunks in the system.

The right pane represents Endpoints for the chosen Gateway/Trunk or all Endpoints.

Icons of the Gateways, Trunks and Endpoints change their color according to their current status.

The color of the Gateway and Endpoint status icon indicates a status according to Table E-1.

The color of the Trunk Status icon indicates a status according to Table E-2.

Table E-1: Gateway and Endpoint Status Icon

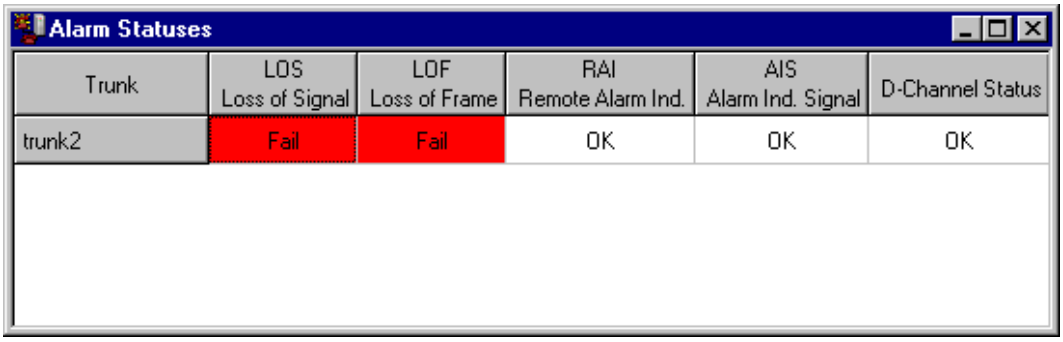
Icon Color	Gateway and Endpoint Status
Green	"Active" state; connections are possible.
Red	"Disconnected" state; there is no response from the Gateway/Endpoint.
Green & Red	Can be seen for E1/T1 trunk gateways only; the Gateway is connected but at least one of its trunks has an Alarm or an unsynchronized D-channel.
Blank	"Disabled" state; Gateway/Endpoint manually disabled by the User.

Table E-2: Trunk Status Icon

Icon Color	Trunk Status
Red	There's an alarm on the trunk, or else PRI D-channel is not synchronized (for PRI ISDN only).
Green	The trunk can be used for calls. There is no alarm and PRI D-channel is synchronized.

To view an alarm's details, select the Trunk, right-click and choose the option "Show Alarm Status"; the Alarm Status window opens:

Figure E-3: Alarm Statuses



Note: The option is only available when the program is in Active state.

➢ To obtain the actual status of Gateways, Trunks and Endpoints, take the following step:

1. Manually refresh the Gateways & Endpoints Configuration window (shown in Figure E-2 on page 66) by pressing F5 or by selecting the option "Refresh" from the "View" menu.

➢ To view Endpoints (in Phone # or Gateway sequence), take the following step:

1. Click on the appropriate column's title.

All changes to the database are saved in cache memory. When the application exits, data is saved on the hard disk.

➢ To save data from the cache memory on the hard disk, perform EITHER step 1 OR step 2:

1. Click the icon Save to Database in the Gateways and Endpoints Configuration window (shown in Figure E-2 on page 66).

2. Select the option "Save to DB" on the Gateways menu in the Gateways and Endpoints Configuration window.

A backup of the current database can also be created as a file. The option is available from Call Manager's main window (shown in Figure E-1) on the "File" menu. The backed-up database can be restored from the file at any time.

Note: The restored database fully replaces the current one, so it's recommended to back up the current database before using the "Restore" option.

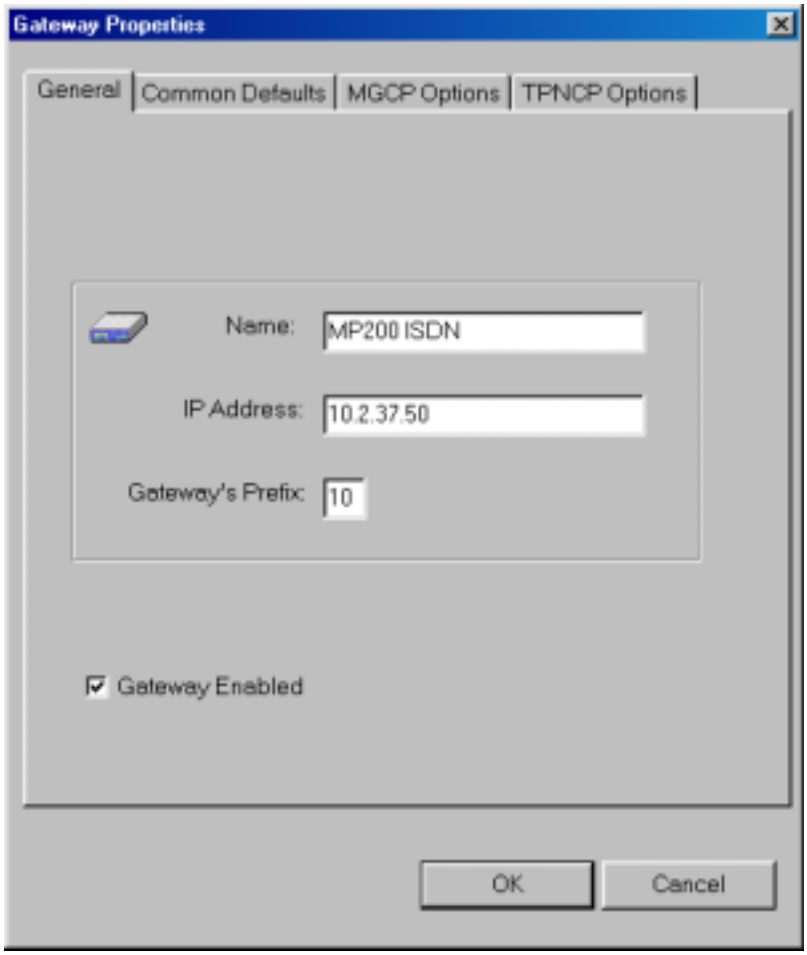
E.2.2 Gateway Configuration

➢ To add/remove/edit a Gateway:

1. Click the "Add Gateway", "Edit Gateway" or "Remove Gateway" icons on the Toolbar of the Gateways and Endpoints Configuration window (shown in Figure E-2 on page 66). The same options are on the "Gateways" menu.

2. For each Gateway, specify the name and the IP address of the Gateway.

Figure E-4: Gateway Properties - General



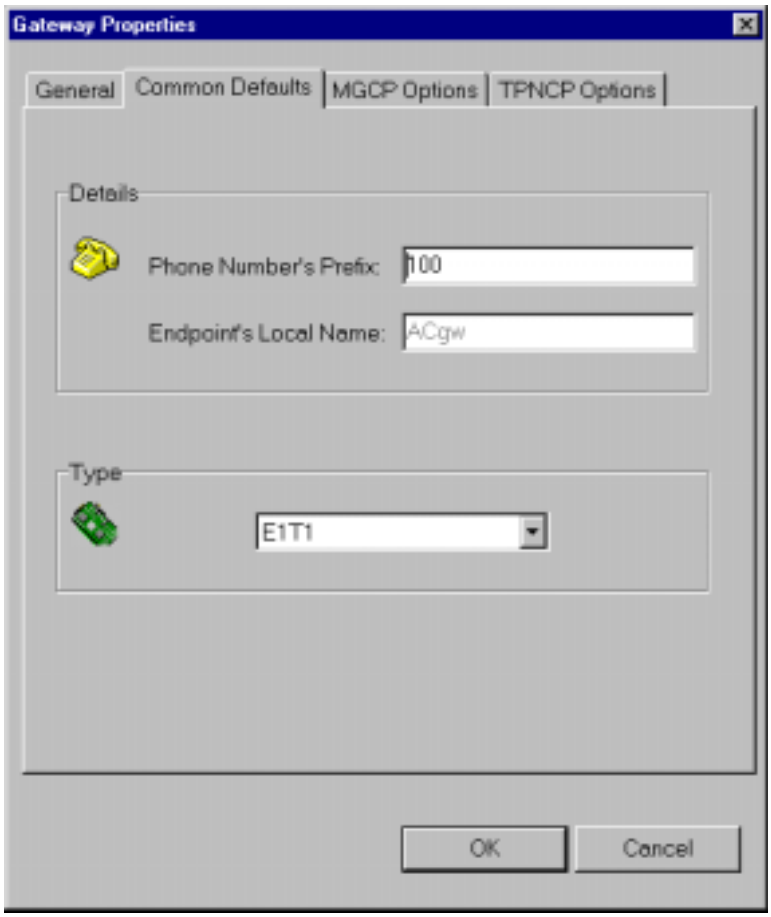
"Gateway's Prefix" number can be used to obtain a random line of all Gateways having the same Gateway prefix.

Clearing the "Gateway Enabled" check box disables the Gateway. A disabled Gateway doesn't participate in the activity of Call Manager.

The Gateway configuration also contains the default parameters of its Trunks and Endpoints.

Select the "Common Defaults" tab to set the common default parameters:

Figure E-5: Gateway Properties – Common Defaults



The parameters in the window are divided into 2 categories:

1. **Details**

   Phone Number's Prefix - the common phone number's prefix for the Endpoints of the Gateway.

   Endpoint's Local Name - the common prefix for the Endpoints' local names.

2. **Type**

   Select FXS, FXO or Transparent default according to Gateway type.

   Select FXS for Endpoints that are connected to phones; select FXO for Endpoints connected to PBX or to PSTN.

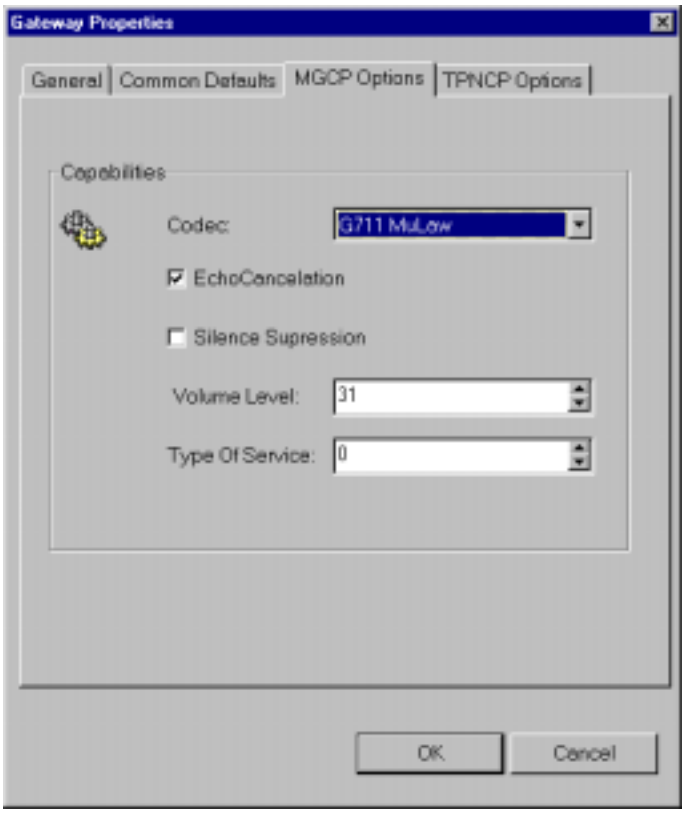
   Transparent type can be defined for any Endpoint. A transparent Endpoint doesn't support any signaling. To create connections to or from a transparent Endpoint, see Section 0 (E.2.4.1 Transparent Type Specific Configuration) on page 81. Transparent type can be used for testing the MP-200 E1/T1 trunk Endpoints.

   Select E1/T1 type for E1/T1 trunk gateways that support TPNCP (AudioCodes proprietary) protocol.

   For a Gateway that contains mixed Endpoints, enter the Endpoint configuration window and select the correct Endpoint type.

Select the “MGCP Options” tab to set the MGCP default parameters:

Figure E-6: Gateway Properties – MGCP Options



Parameters in the Gateway Properties window are:

Codec: the default voice compression coder.

Echo Cancelation is ON when the check box is checked.

Silence Suppression is ON when the check box is checked.

Volume Level (should be in the range 0-63).

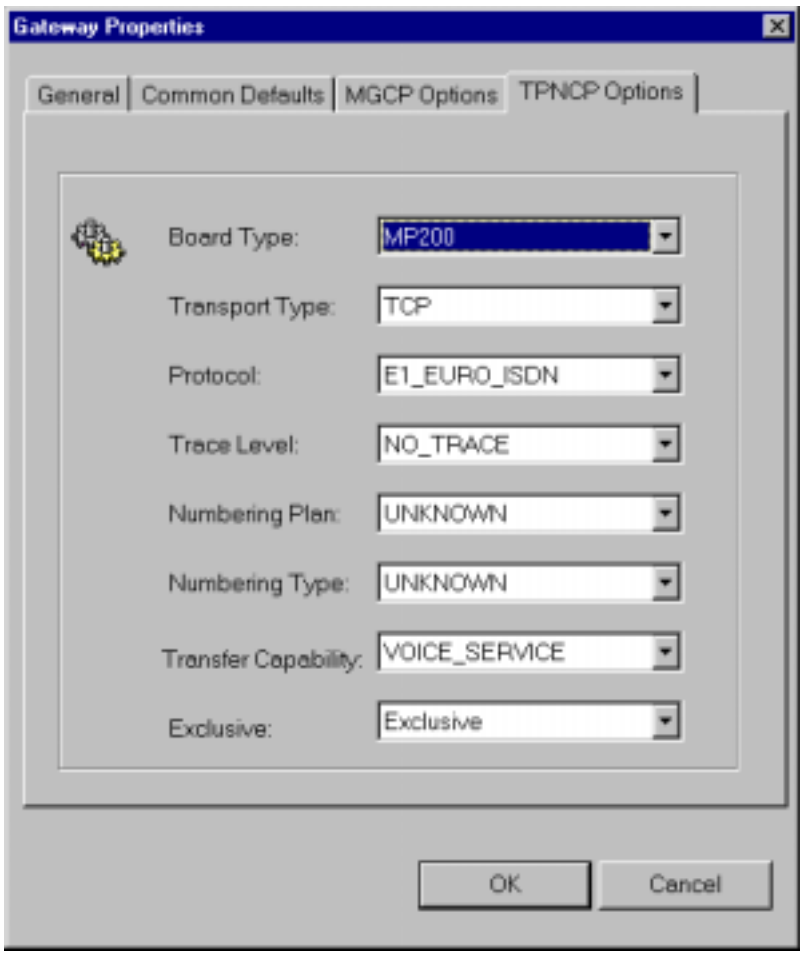
Type of Service for IP Datagram (should be in the range 0-255).

The scope of capability parameters applies to all Gateway Trunks and Endpoints selected to use “*Gateway’s Default Capabilities”.*

Gateway capabilities can be changed during the operation of Call Manager.

Select the "TPNCP Options" tab to set the TPNCP default parameters (for E1/T1 type only):

Figure E-7: Gateway Properties – TPNCP Options



Parameters in the Gateway Properties window used for gateway initialization are:

- Type of board.
- Transport type (UDP or TCP).
- Trunk protocol. Also defines trunk's type – E1 or T1.
- Trace Level – set for debugging purposes, according to the TPNCP manuals.

The other parameters are used to configure PRI ISDN Setup message.

The scope of these parameters applies to all Gateway Trunks selected to use "*Gateway's Default Capabilities".*

Note 1: Useful parameters of the above fields, for Media Gateway modules, are Board Type and Transport Type.

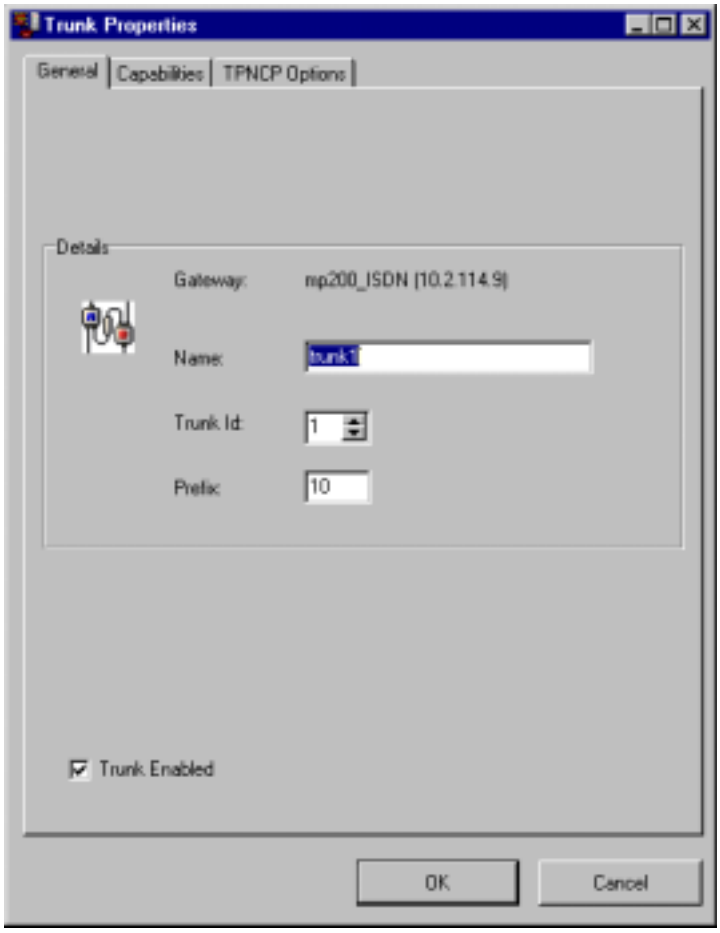
Note 2: Changes to the above parameters are not applied after the operation of Call Manager.

E.2.3 Trunk Configuration

➢ To add/remove/edit the Trunks:

1. Select the host Gateway and use the buttons in the Gateway's pane. The same options are also available from the menu "Trunks".
2. For each Trunk, specify the name and the Trunk ID according to the physical trunk ID number.

Figure E-8: Trunk Properties - General



"Prefix" can be used to obtain a random line of the specified Trunk.

The "Capabilities" tab is identical to the "MGCP Options" tab in the Gateway definition window. It inherits all data from the Host gateway. The User can change the parameters to be unique per trunk.

The "TPNCP Options" tab is identical to the "TPNCP Options" in the Gateway definition window.

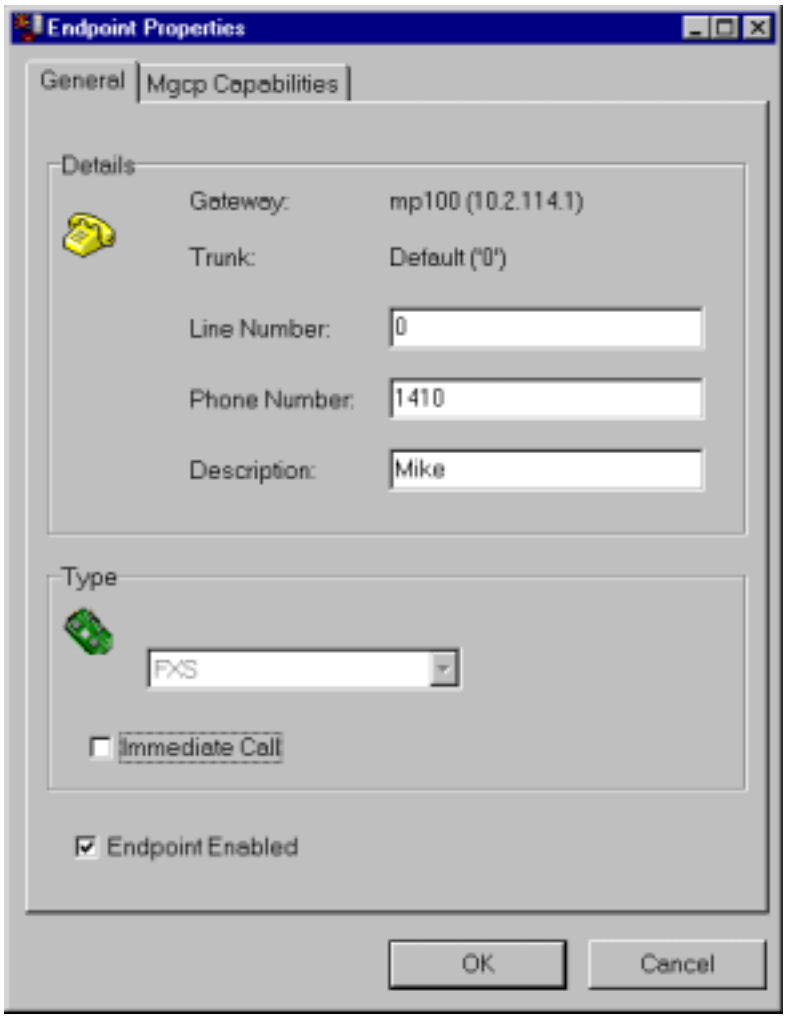
E.2.4 Endpoint Configuration

For each Gateway or Trunk, a number of Endpoints can be specified.

➢ To add an Endpoint for the Gateway:

1. Select one of the Gateways or Trunks in the Gateway's tree.
2. Press the "Add Endpoint" button or select from the "Endpoints" menu the "Add" option.

Figure E-9: Endpoint Properties – General



The following general Endpoint Properties can be defined (as shown in Figure E-9).

1. ***Line Number*** is the Gateway port number (0 to 7 for MP-100).

2. ***Phone Number*** is a string of digits (default is 4 digits). The number should be unique for Call Manager. Digit map can be configured via "Configuration" window, "Telephony" tab.

3. ***Description*** is a free description of the endpoint (like a name of the line's owner).

4. ***Type*** is the Endpoint type (FXS/FXO/Transparent/E1/T1).

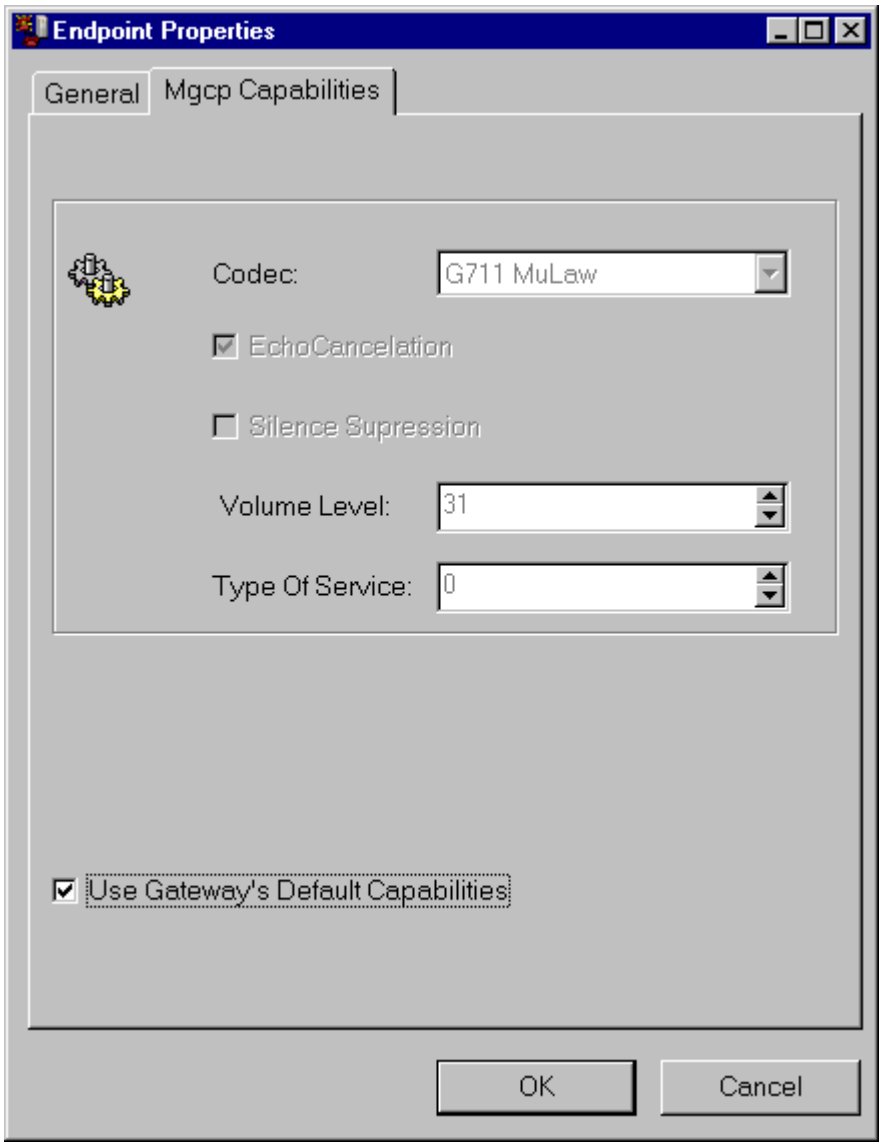
5. ***Immediate Call*** – when selected, "Dest Phone Number" field should be a valid phone number, belonging to one of the Endpoints controlled by Call Manager (gateway's prefixes and random calls prefixes are possible too). When the defined endpoint is the initiator of a call, the specified phone number is dialed automatically. *Endpoint Enabled* – if not selected, the Endpoint is disabled.

Note 1: Sequence of Endpoints can be defined as an interval of channels in the field "Line Number".
For example, in the above window, the "Line Number" field is defined as 0-2, and all the other fields are unchanged.
Instead of one Endpoint, three Endpoints are created:
channels: 0,1,2; telephone numbers: 1450, 1451, 1452.
Select "Type" and all the other parameters stay as defined in the original window.

Note 2: To change an Endpoint's type, disable the endpoint, set the new type, and then enable the Endpoint.

Select the “Capabilities” tab in order to set Endpoint capabilities:

Figure E-10: Endpoint Properties – MGCP Capabilities



The following MGCP Capability Endpoint Properties can be defined (as shown in Figure E-10):

1. *Codec* is the voice compression coder type.

2. *Echo Cancellation* is ON when the check box is checked.

3. *Silence Suppression* is ON when the check box is checked.

4. *Volume Level* is an Endpoint gain (should be in the range 0-63).

5. *Type of Service* is an indication of the quality of service requested for an IP datagram (should be in the range 0-255).

6. *Use Gateway's Default Capabilities* - if selected, the Endpoint capabilities are defined by the Gateway/Trunk MGCP Defaults; otherwise, select individual capabilities for each Endpoint.

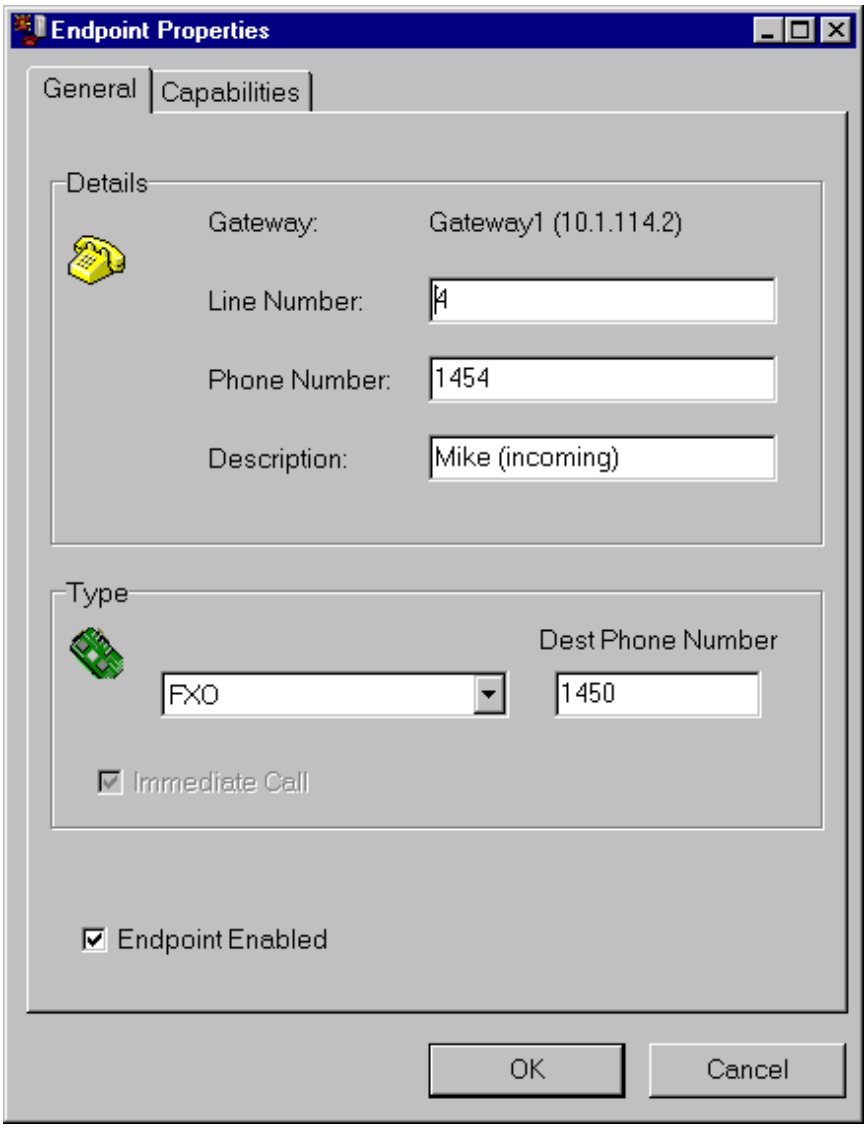
Note 1: If "Use Gateway's Default Capabilities" is checked, then the Endpoint "Capabilities" parameters cannot be edited.

Note 2: Endpoint capabilities can be changed during Call Manager operation.

FXO Type Specific Configuration

For the FXO Endpoint, define the Destination phone number. This number will be used for routing the **incoming** Call.

Figure E-11: FXO Endpoint Specific

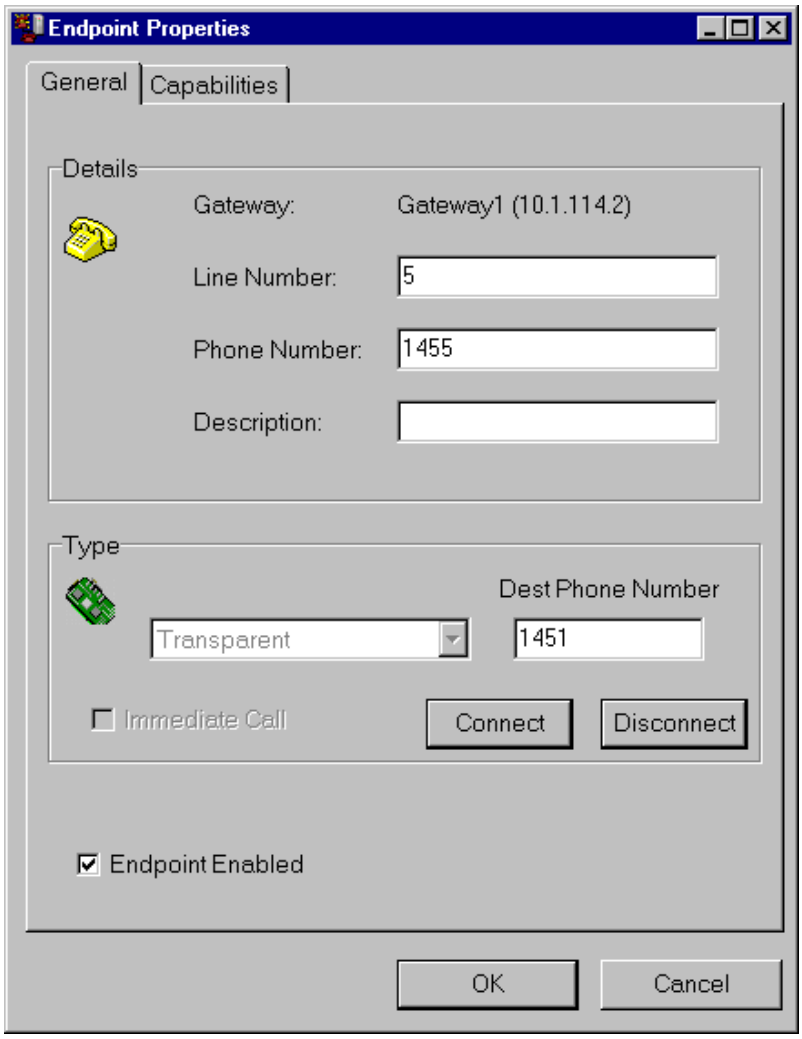


The "Destination Phone Number" should be a valid prefix or FXS/FXO Endpoint phone number, belonging to one of the Endpoints controlled by Call Manager.

E.2.4.1 Transparent Type Specific Configuration

For Transparent Endpoint, define the Destination phone number. This number will be used for routing the outgoing Call.

Figure E-12: Transparent Endpoint Specific



"Dest Phone Number" should be a valid FXS or Transparent phone number belonging to one of the Endpoints controlled by Call Manager.

Note: Buttons “Connect” and “Disconnect” are enabled only when Call Manager is activated.

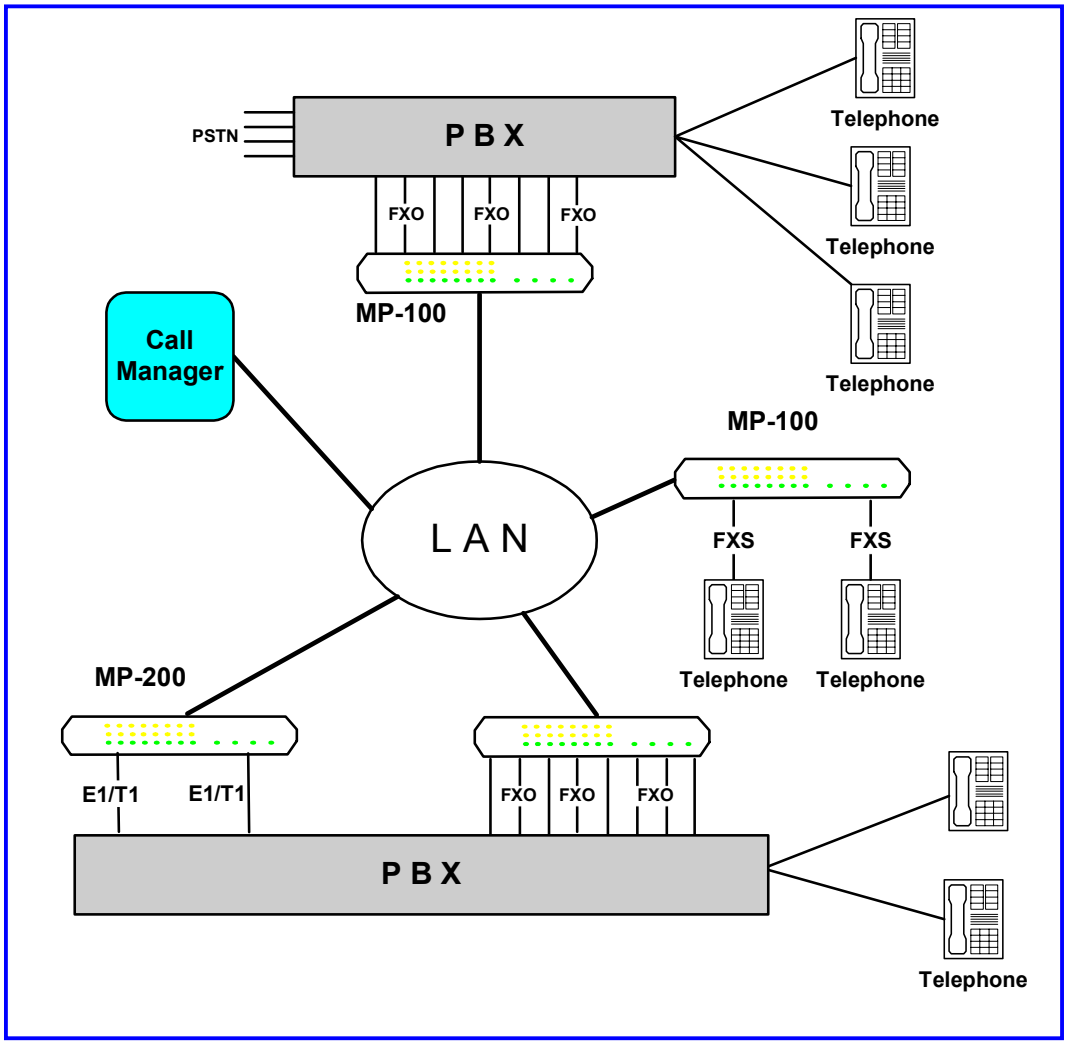
E.2.4.2 E1/T1 Type Specific Configuration

For E1/T1 Endpoint, the User can define the Destination phone number. This number will be used as destination in ISDN Q.931 setup message.

E.3 Calls Supported by Call Manager

Call Manager supports the following calls configuration scheme:

Figure E-13: Supported Calls



E.3.1 FXS *to/from* FXS Call

To make a call from one FXS Endpoint to another, dial the phone number of the destination FXS Endpoint. If it is valid and not busy, the conversation will be established.

E.3.2 FXO *to/from* FXO Call

The Endpoint incoming call will be transferred by Call Manager to the FXO "Destination Phone Number" defined for the Endpoint. The type of the call can be used for connecting two PBX units via IP.

E.3.3 FXO *to* FXS Call (incoming call)

The Endpoint incoming call will be transferred by Call Manager to the "Destination Phone Number" defined for the FXO Endpoint.

E.3.4 FXS *to* FXO Call (outgoing call)

Four ways to make a call from an FXS Endpoint to the FXO Endpoint connected to PBX:

1. Dial '9' from the FXS Endpoint (the User can set another number for the call's type via the "Telephony" tab in the "Call Manager Configuration" window). Call Manager will select a **not busy** FXO line. If all FXO Endpoints are busy, a busy tone will be heard.
2. Dial the phone number **of the specific** FXO Endpoint from the FXS Endpoint. If the Endpoint is not busy, the conversation will be established through the Endpoint.
3. Dial '8' from the FXS Endpoint (the User can set another number for the call's type via the "Telephony" tab in the "Call Manager Configuration" window). Call Manager will search for the FXO Endpoint for which the "Destination Phone Number" is the Phone Number of the originating FXS Endpoint. If such an Endpoint is found and it's not busy, the conversation will be established through this Endpoint.
4. Dial the "prefix" of some FXO gateway. Call Manager will select a line that isn't busy from the gateway with the dialed prefix.

E.3.5 Transparent *to/from* Transparent Call

A Transparent Call is a call without signaling. To make a call, open the "Endpoint Properties" window of the source Endpoint (see Section E.2.1, "Gateways & Endpoints Configuration", on page 66). Press the "Connect"

button; you'll be connected to the phone number defined in the "Destination Phone Number" field. Note that the destination line should be a valid Transparent Endpoint that is not busy.

To finish the call, press the "Disconnect" button in the "Endpoint Properties" window of the source or destination Endpoint.

E.3.6 Transparent *to* FXS Call

Making a call from a Transparent Endpoint to an FXS Endpoint is the same as Transparent to Transparent. On the FXS Endpoint, the usual signaling is applied. When the "Connect" button is pressed on the telephone connected to the destination FXS Endpoint, a ringing tone is heard.

The call is finished when the destination FXS phone is on-hooked, or the "Disconnect" button is pressed in the "Endpoint Properties" window of the Transparent Endpoint.

E.3.7 FXS *to* Transparent Call

To make a call from an FXS Endpoint to a Transparent Endpoint, dial the phone number of the destination Transparent Endpoint. If it is valid and not busy, the conversation will be established.

The call is finished when the FXS Endpoint's telephone is on-hooked, or the "Disconnect" button is pressed in the "Endpoint Properties" window of the Transparent Endpoint.

E.3.8 Selecting All Transparent Endpoints

Note: The "Gateways & Endpoints Configuration" window features options to control connections for all Transparent Endpoints defined in the system. This saves you from having to open the "Endpoint Properties" window for each of the Transparent Endpoints and pressing the "Connect" or "Disconnect" button, for each selection.
The options are available from the "Gateways" menu in the "Gateways & Endpoints Configuration" window and the "All" key selected in the Gateway's tree.

E.3.9 FXS *to* E1/T1 Call (outgoing call)

➢ To make a call from an FXS Endpoint to the E1/T1 Trunk connected to PBX, you can do either 1, 2, or 3:

1. Dial the phone number of a specified B-channel (E1/T1 Endpoint phone number). Call Manager will check that the selected B-channel is valid and the User will get the second dial tone. The second dialed number will be set as Destination in the ISDN Q.931 setup message.
2. Dial the "prefix" of some gateway or trunk. Call Manager will first select the **not busy** B-channel (E1/T1 Endpoint) of the specified Gateway/Trunk, and the User will get the second dial tone as before.
3. Define the destination phone number for an ISDN Q.931 setup message manually per B-channel; the second dial tone will not be heard. To manually define the destination, select the "Immediate Call" check box of the E1/T1 Endpoint and in the "Destination Phone Number" field, set the ISDN destination.

E.3.10 E1/T1 *to* FXS or FXO Call (incoming call)

Destination FXS phone number is retrieved from the incoming ISDN Q.931 setup message.

E.3.11 E1/T1 *to* E1/T1 Call (tandem call)

"Destination" filed in the incoming ISDN Q.931 setup message must be a valid E1/T1 Endpoint phone number or one of the Gateway/Trunk prefixes defined in the system.

E.4 Call Manager Activation/Deactivation

To activate the program, press the "Start" button in Call Manager's main window, or select the option "Start Call Manager" from the File menu.

To deactivate Call Manager, press the "Stop" button or select the option "Stop Call Manager" from the File menu.

When Call Manager has stopped, it tries to delete any connections that still exist on some Endpoints. The feature can be disabled via the "Configurations" window ("General" tab). If the feature is disabled when the application is closed, some RTP packages could possibly still be on the network. To stop sending the packages, the MGCP command DLCX must be executed on each sending Endpoint.

In the application, there is an option to send a DLCX command to all Endpoints of the selected Gateway. The option is available in the "Gateways & Endpoints Configuration" window. Select one of the listed Gateways (the "All" option can also be used), right-click and choose the "Delete Connections" option.

E.4.1 Debug Window

If the Log Messages are selected, they will appear in the Debug dialog. The window is opened together with the main window of the application.

To clear messages in the Debug dialog, choose the "Tools" menu and then the "Clear Debug Messages" option.

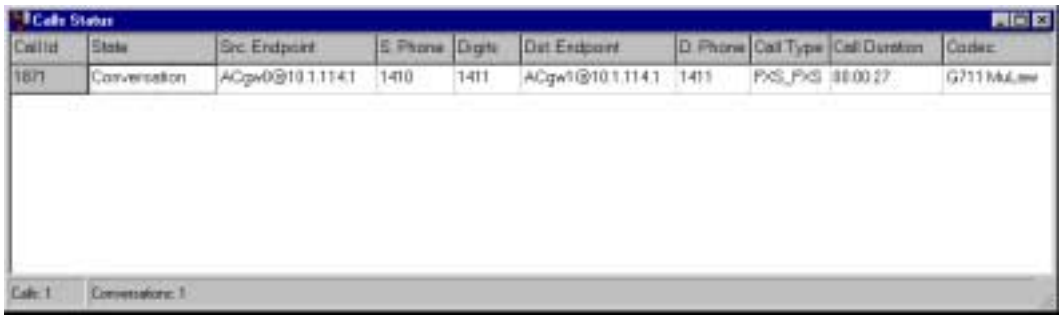
If the window is closed, it can be reopened by selecting the option "Debug Messages" from the "View" menu.

To temporary stop messages in the Debug window, right-click in the main application window. In the open menu, choose the option "Stop Debug Messages". To enable the messages again, choose the option "Run Debug Messages" in the same menu as before.

E.4.2 Call Status

View the status of current Calls by pressing the “Calls” button or choosing the “Calls Status” option in the “View” menu.

Figure E-14: Calls Status



Run-time information on the calls is displayed in the above window.

To stop displaying data in the window, right-click and choose the option “Freeze” in the Open menu.

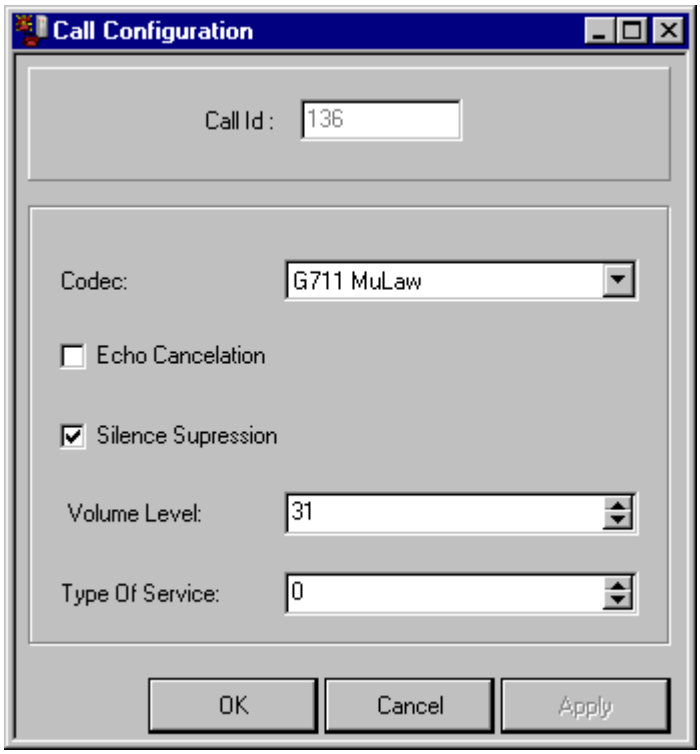
To run the window again, right-click and choose the option “Run” in the Open menu.

The Status bar displays the number of open calls and how many calls are in “Conversation” state (voice can be passed).

E.4.3 Call Configuration

For Calls in **"Conversation" state,** a call can be reconfigured in run time by clicking on "Call Id".

Figure E-15: Call Configuration



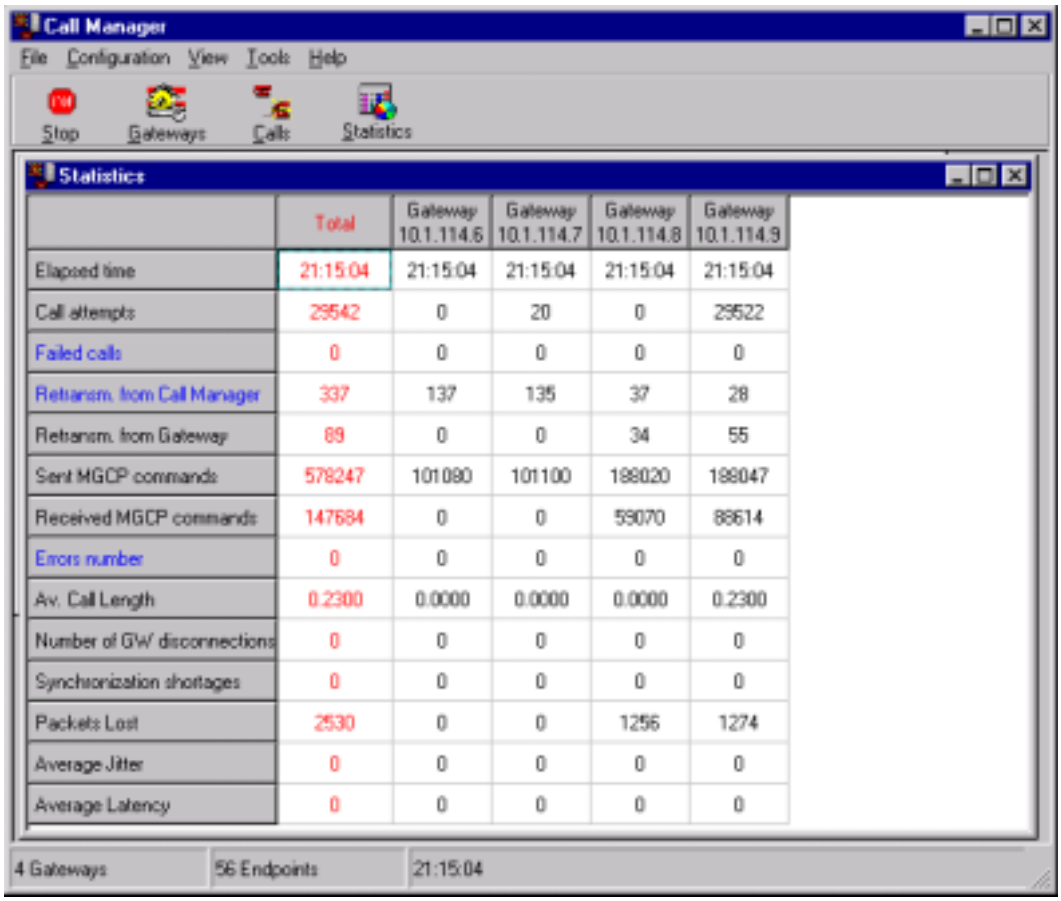
All parameters in the above window (except "Call Id") can be changed. By pressing "Apply" or "OK" button, all changes will be temporally applied to both endpoints participating in the call.

If the configured Call is finished, its "Configuration" window will be closed automatically.

E.4.4 Statistics

To view the statistics parameters of the current session, click the “Statistics” button or choose the “Statistics” option in the “View” menu.

Figure E-16: Statistics



	Total	Gateway 10.1.114.6	Gateway 10.1.114.7	Gateway 10.1.114.8	Gateway 10.1.114.9
Elapsed time	21:15:04	21:15:04	21:15:04	21:15:04	21:15:04
Call attempts	29542	0	20	0	29522
Failed calls	0	0	0	0	0
Retransm. from Call Manager	337	137	135	37	28
Retransm. from Gateway	89	0	0	34	55
Sent MGCP commands	578247	101080	101100	188020	188047
Received MGCP commands	147684	0	0	59070	88614
Errors number	0	0	0	0	0
Av. Call Length	0.2300	0.0000	0.0000	0.0000	0.2300
Number of GW disconnections	0	0	0	0	0
Synchronization shortages	0	0	0	0	0
Packets Lost	2530	0	0	1256	1274
Average Jitter	0	0	0	0	0
Average Latency	0	0	0	0	0

Each statistic counter is per Gateway. “Total” is a sum or an average of the Gateway’s counters. Sometimes while updating statistic parameters, it’s impossible to know to which Gateway the event belongs. In this case, only the “Total” counter is updated.

Automatic reset of all data in the window is performed when Call Manager is activated.

All counters are updated every second.

To stop displaying data in the window, right-click and choose the option "Freeze" from the open menu.

To run the window again, right-click and choose the option "Run" from the open menu.

To reset counters of a specific Gateway, or counters of all defined Gateways, in run time, right-click on the title of the Gateway to reset (or "Total" to reset all) and choose the option "Reset Gateway Counters" from the open menu.

Description of Statistic parameters:

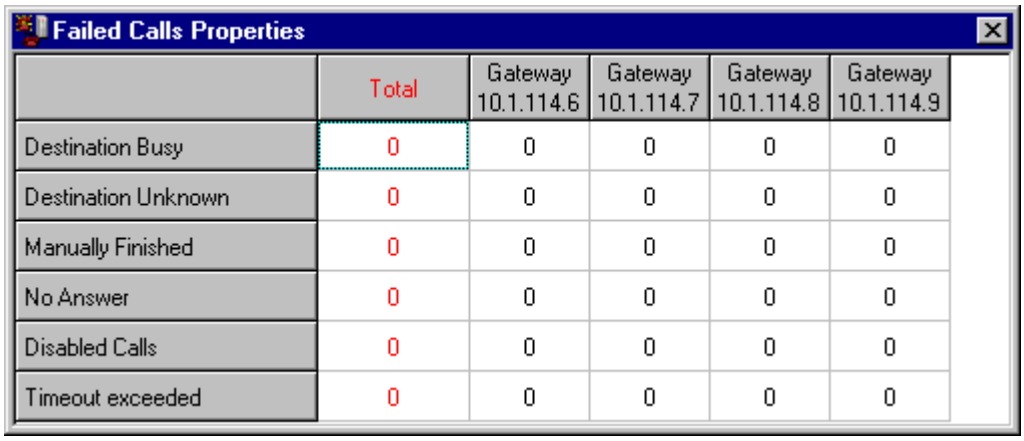
- *Elapsed Time* - time elapsed from last reset.
- *Call Attempts* - number of attempts to call.
- *Failed Calls* - number of unsuccessful calls (conversation hasn't been established).
- *Retransmissions from Call Manager* – the number of times some MGCP command or response was retransmitted by Call Manager.
- *Retransmissions from Gateway* – the number of times some MGCP command or response was retransmitted by Gateway.
- *Sent MGCP commands* - total number of sent MGCP commands.
- *Received MGCP commands* - total number of received MGCP commands.
- *Errors number* - total number of errors in the system.
- *Av. Call Length* - average length of Call per Gateway and per the system.
- *Number of GW disconnections* – the number of times Call Manager lost communication with Gateway.
- *Synchronization shortages* – the number of times there was no synchronization between the endpoint's state inside Gateway and the endpoint's state in the internal Call Manager's state machine.
- *Packets Lost* - total number of lost packets.
- *Average Jitter* - average estimate of the statistical variance of the RTP data packet interval time measured in milliseconds.
- *Average Latency* - average estimate of the network latency in milliseconds.

To obtain properties for some of the above parameters (having blue titles in Figure E-16), click on the chosen title; the "Properties" window opens.

This option is available for all parameters shown in Figure E-17, Figure E-18

and Figure E-19.

Figure E-17: Failed Calls Properties

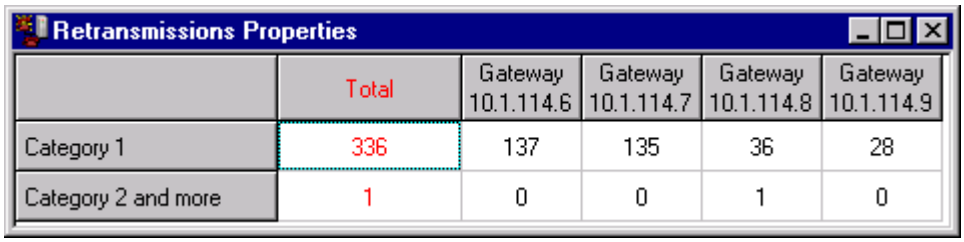


The window presents the types of failed calls, each with a counter.

Description of the above parameters:

- Destination Busy - destination endpoint is busy.
- Destination Unknown - not a valid destination phone number.
- Manually Finished - the call is interrupted before dialing a phone number.
- No Answer - no answer at the destination.
- Disabled Calls - one of the endpoints participating in the Call is manually disabled (via the “Gateways & Endpoints Configuration” window).
- Timeout exceeded – the Call is terminated when timeout is exceeded in one of the states of the call’s state machine.

Figure E-18: Retransmission Properties



	Total	Gateway 10.1.114.6	Gateway 10.1.114.7	Gateway 10.1.114.8	Gateway 10.1.114.9
Category 1	336	137	135	36	28
Category 2 and more	1	0	0	1	0

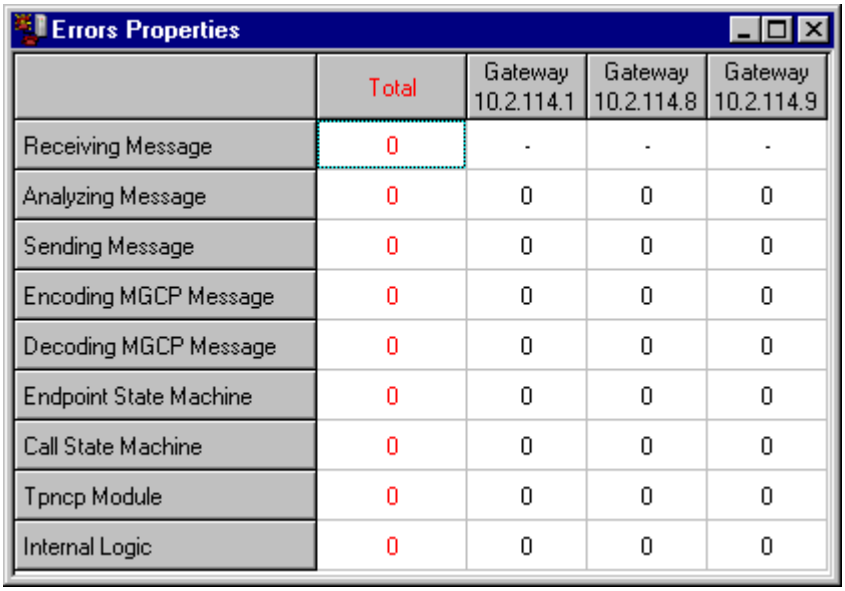
For each MGCP command sent, Call Manager expects to receive an acknowledgment.

If there is no acknowledgment within the defined period of time (this can be set in the "Call Manager Configuration" window), the same command is retransmitted once. The next retransmission will always be after twice the previous period of time.

The above window shows the number of commands sent within each period.

- *Category 1* – the number of commands sent during the first retransmission.
- *Category 2 and more* – the number of commands sent after the first retransmission.

Figure E-19: Error Properties



	Total	Gateway 10.2.114.1	Gateway 10.2.114.8	Gateway 10.2.114.9
Receiving Message	0	-	-	-
Analyzing Message	0	0	0	0
Sending Message	0	0	0	0
Encoding MGCP Message	0	0	0	0
Decoding MGCP Message	0	0	0	0
Endpoint State Machine	0	0	0	0
Call State Machine	0	0	0	0
Tpncp Module	0	0	0	0
Internal Logic	0	0	0	0

All error messages in the system are classified according to the following list.

Error classification:

- Receiving Message - an error in the network socket or in the thread that manages incoming messages.
- Analyzing Message - unexpected MGCP command's or response's format.
- Sending Message - an error in the network socket or in the thread that manages outgoing messages.
- Encoding MGCP Messages - an error while composing an MGCP message to send.
- Decoding MGCP Message - parser error.
- Endpoint State Machine - received unexpected event for a current state of Endpoint's State Machine.
- Call State Machine - received unexpected event for a current state of Call's State Machine.

- TPNCP Module – an error in the VoIPLib (error event is received) or an error while executing VoIPLib command.
- Internal Logic - all other possible errors (memory management, data management, etc.). Usually such errors are critical.

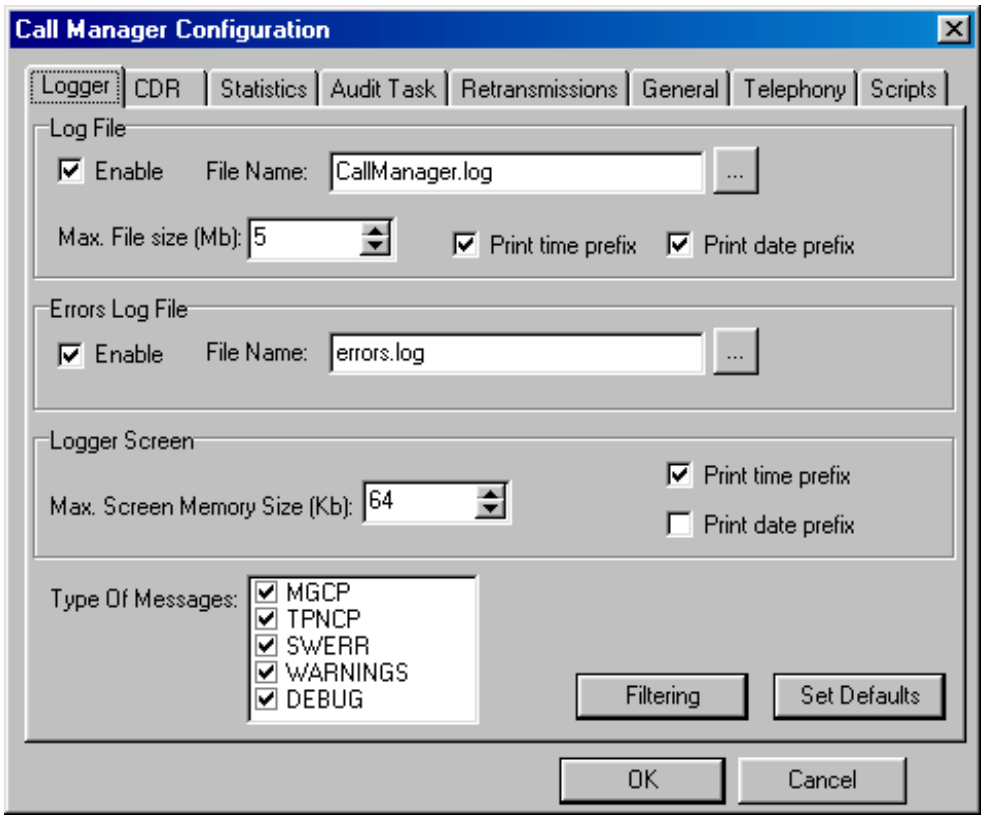
E.5 Call Manager Configuration

Select "Configuration" and then "Call Manager"

E.5.1 Logger

The selection enables configuring the application's Log Messages.

Figure E-20: Logger Configuration



Configuring the *Log File* of the application.

- If the check box “Enable Log File” is checked, all Log messages are saved in the log file, the name being specified in the text field “File Name”.
- When the size of the Log File reaches its maximum (“Max. File size”) a new Log File is started (with new name). For example, if the source Log File’s name was logfile.log, then following files will be logfile_1.log, logfile_2.log and so on.
- Log messages can be saved with the current date and time.

Configuring the Errors Log File of the application.

- If the check box “Enable Errors Log File” is checked, all Log messages of type ‘SWERR’ or ‘WARNINGS are saved in the file, the name being specified in the text field “File Name”. The same messages can be found inside the regular Log File.

Configuring the Logger window - displays Log messages in the “Debug” window.

- The maximum number of displayed messages can be managed by changing the “Max. Screen Memory Size” parameter.
- Log messages can be saved with the current date and time.

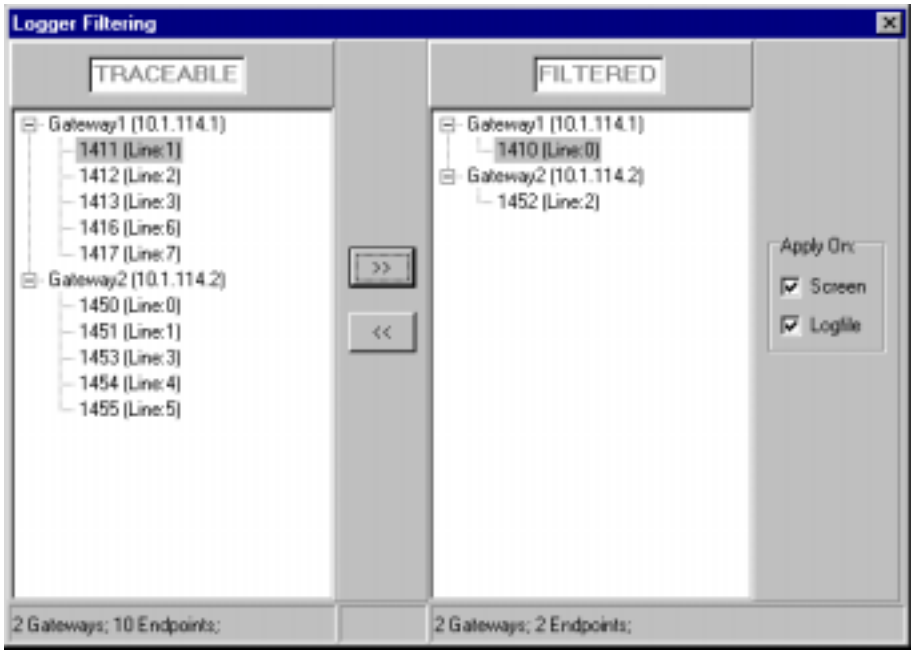
Five types of log messages:

Log Message Type	Description
MGCP	Commands of the Media Gateway Control Protocol.
TPNCP	Commands of TPNCP.
SWERR	Unsupported events in the program’s state machines or internal logic errors.
WARNINGS	Notifications on communication problems and failed calls.
DEBUG	Different debug messages of the program.

E.5.2 Message Filtering

The User can cancel Log Messages for some gateways/trunks/endpoints by pressing the "Filtering" button.

Figure E-21: Logger Filtering



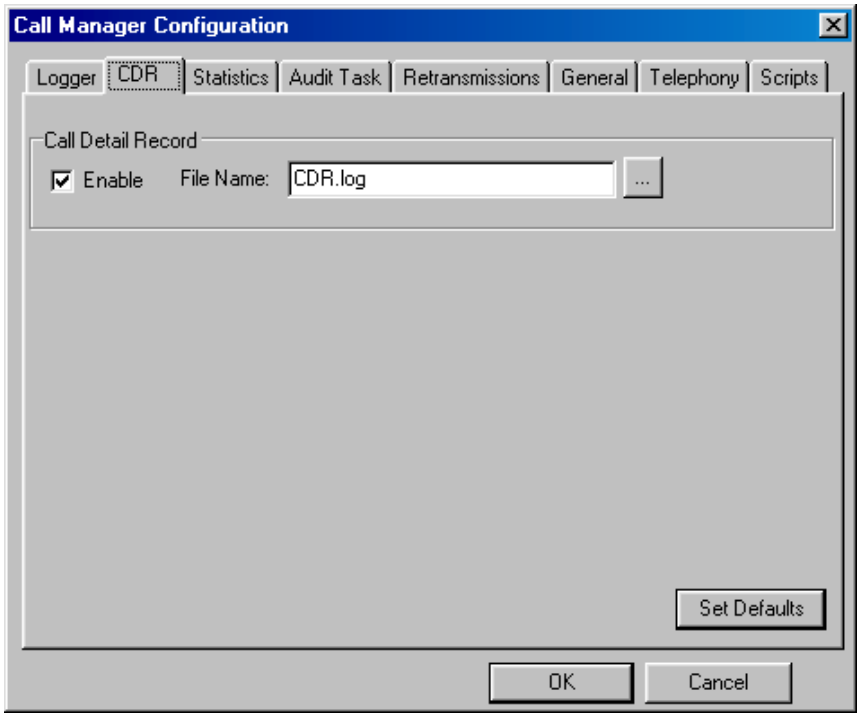
The two-pane window enables Log Messages filtering. The left pane displays all **unfiltered** gateways and endpoints. The right pane displays all **filtered** gateways and endpoints. To move gateways/endpoints between the two panes, use the buttons with arrows.

The filter can be applied for messages printed on the screen, into the Log file, or both. The feature can be managed by checking the "Apply On" check boxes.

E.5.3 CDR (Call Detail Recording)

Select the CDR tab:

Figure E-22: Call Detail Recording



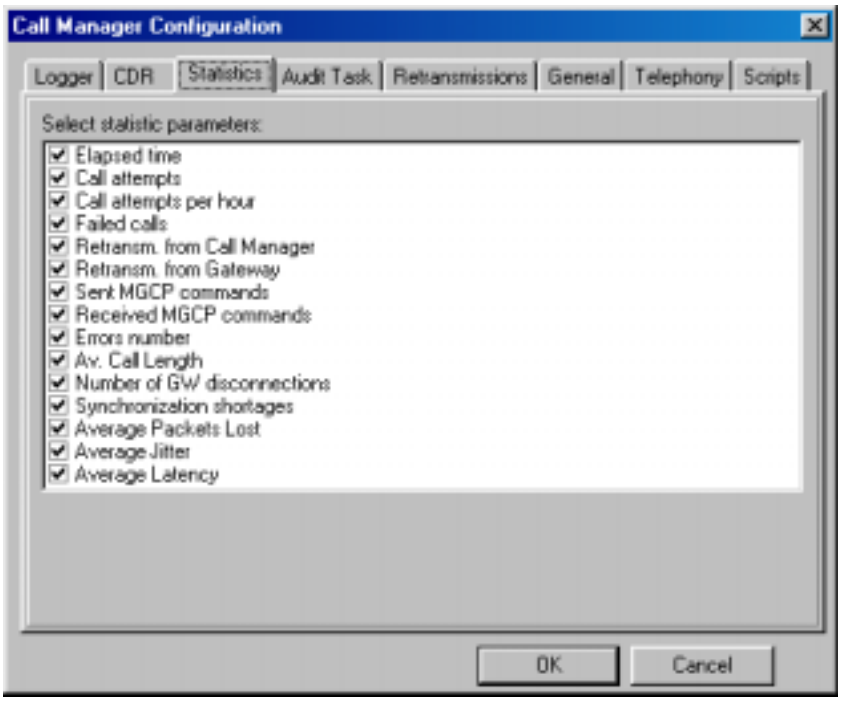
The CDR file contains the following information for each Call:

```
CallId
Date & Time
Call Type
Originating and Destination Endpoints
Dialed Number
Answered Time
Duration of the Call in seconds
Originating endpoint number
Originating Connection Id
Destination endpoint number
Destination Connection Id
Termination reason
Codec
Origination Packet Lost
Destination Packet Lost
```

E.5.4 Statistics

Select the "Statistics" tab:

Figure E-23: Statistics Configuration

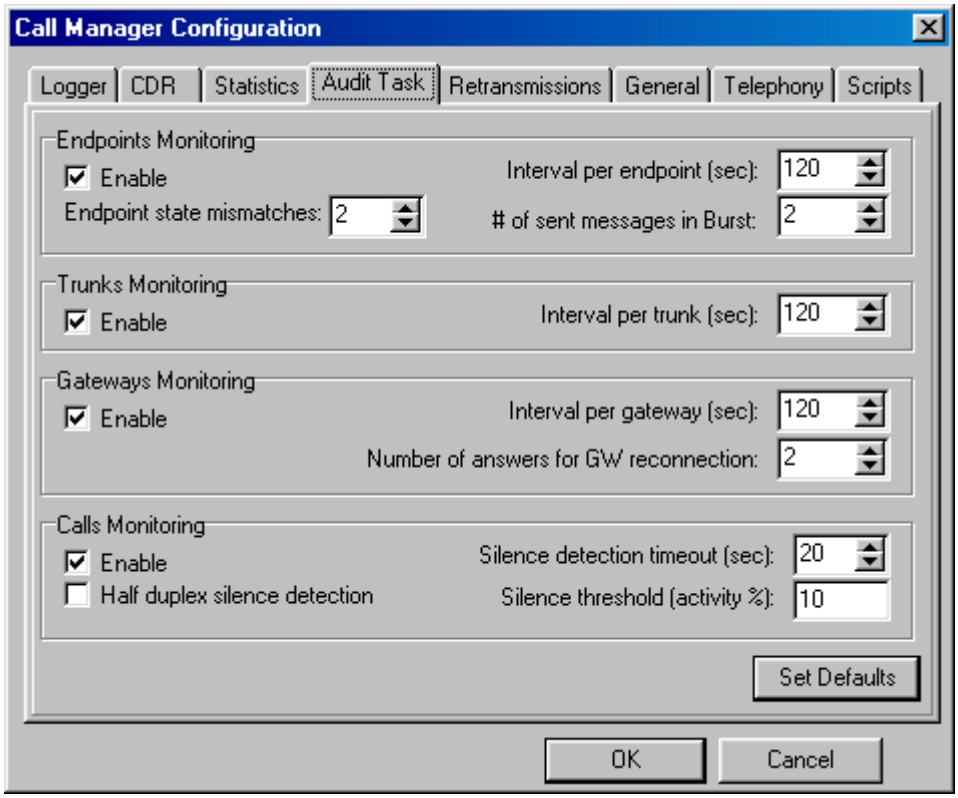


Use the check boxes in the above window to determine what statistic parameters will be displayed in the "Statistics" window.

E.5.5 Audit Task

Select the “Audit Task” tab:

Figure E-24: Audit Task Configuration



E.5.5.1 Endpoints Monitoring Configuring

The task works for MGCP endpoints only. This means it does not work for E1/T1 type.

If the check box “Enable” is checked, then the Endpoints Monitor will be active.

The purpose of the task is synchronization between logical and actual endpoints’ states and lists of requested events.

The task continuously sends “AUEP” to all endpoints defined in the system.

To manage the task’s performance, the User can change the following parameters:

- "Endpoint state mismatches" – maximum number of synchronization mismatches. When the specified number of mismatches has occurred for some Endpoint, the logical Endpoint's state is updated (if the Endpoint is participating in some call, then the call will be interrupted).
- "Number of sent messages in burst" – maximum number of AUEP commands sent in one burst (the command is not sent to all defined endpoints at once in order not to overload the network).
- "Interval per Endpoint" – the time interval between sequential AUEP commands per endpoint.

E.5.5.2 Trunks Monitoring Configuring

If the check box "Enable" is checked, the Trunks Monitor will be active.

The purpose of the task is to verify alarm and D-channel statuses of all enabled E1/T1 Trunks.

The task continuously sends acPSTNGetAlarmStatus() and acISDNGetDChannelStatus() (for PRI ISDN protocol only) to every E1/T1 trunk defined in the system.

E.5.5.3 Configuring Gateway Monitoring

If the check box "Enable" is checked, then the Gateways Monitor will be active.

The purpose of the task is to verify connection with all enabled Gateways.

The task continuously sends "AUEP" to one endpoint of every gateway (not E1/T1) defined in the system.

For E1/T1 trunk gateways monitoring, use the TPNCP command: acGetBoardStatus().

To manage a performance of the task, the User can change the following parameters:

- "Interval per Gateway" – the time interval between sequential requests per gateway.
- "Number of answers for GW reconnection" – number of answers from a disconnected Gateway to reconnect the Gateway again.

E.5.5.4 Configuring Call Monitoring

If the check box "Enable" is checked, the Calls Monitor will be active.

The purpose of the task is to detect silence during "FXO to FXO" calls.

The task continuously sends "AUCX" to both connections of the call. Usually, it

can be used when there is no other way to detect the end of a call.

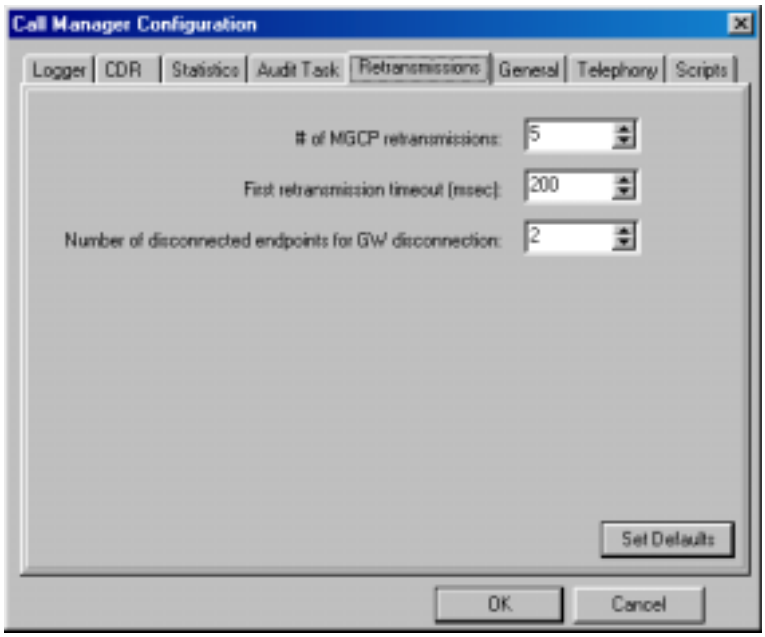
Configuration parameters:

- "Half duplex silence detection" – if checked, silence detection for one of two connections is enough to disconnect the call.
- "Silence detection timeout" – the time interval (in seconds) between sequential AUCX commands per connection.
- "Silence threshold (activity)" – a threshold that defines "silence". It's a percent of the maximum RTP packets flow for the connection.

E.5.6 Retransmissions

Select the "Retransmissions" tab:

Figure E-25: Retransmissions Configuration



Call Manager provides a timeout for all outstanding commands and retries them when timeouts are exceeded. When repeated commands fail to be acknowledged, the host endpoint becomes 'disconnected'.

Configured parameters:

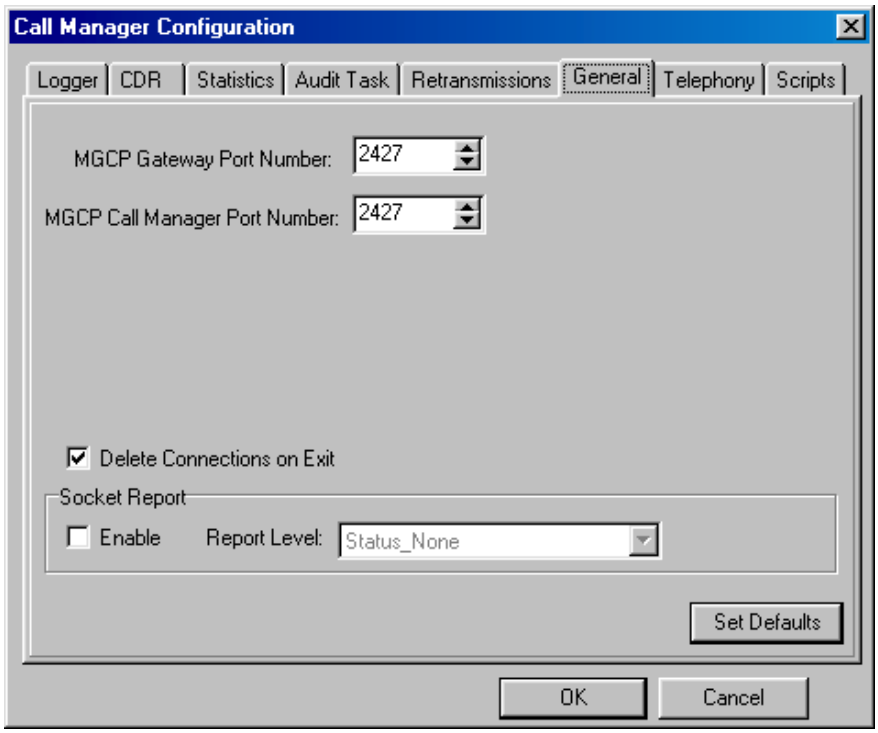
- "First retransmission timeout" - timeout for outstanding command.

- "Number of MGCP retransmissions" - maximum number of retransmissions for outstanding command.
- "Number of disconnected endpoints for GW disconnection" - number of disconnected endpoints that require to be marked that its Gateway is disconnected.

E.5.7 General

Select the "General" tab:

Figure E-26: General Configuration



- "MGCP Gateway Port Number" - number of MGCP port to send messages.
- "MGCP Call Manager Port Number" - number of port to listen to MGCP messages.
- "Delete Connections on Exit" – if the check box is checked and the program is exited or deactivated, any connections that still exist on some Endpoint will automatically be deleted.

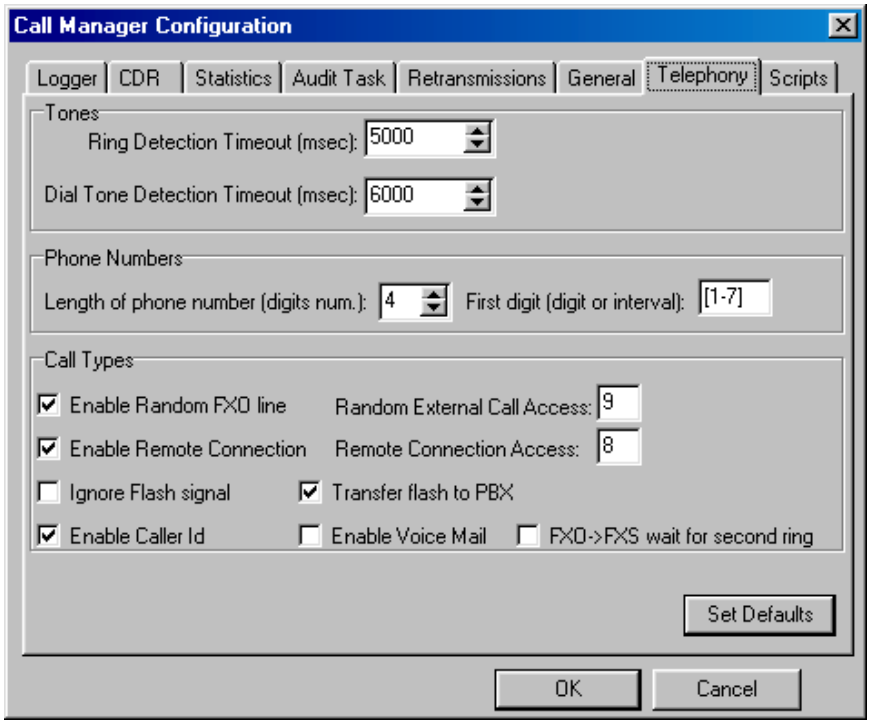
Socket Report group box allows debugger options to be set for the UDP socket:

- To enable the debugger, check the “Enable” check box.
- Choose the debugger’s report level by selecting one of the options in the combo box “Report Level”.

E.5.8 Telephony

Select the “Telephony” tab:

Figure E-27: Telephony Configuration



Tone Parameters:

- “Ring detection timeout” – timeout between sequential ring signals on the FXO endpoint.
- “Dial Tone Detection Timeout” – timeout to detect dial tone on FXO line (not used in version 2.2).

Phone Number Configuring:

- "Length of phone number" – number of digits in the endpoint's phone number.
- "First digit" – first digit phone's number interval. It can be defined as one digit, in which case all phone numbers have to begin with that digit. Alternately, it may be defined as an interval in the format: [from-to].

Call Type Configuring:

- If the "Enable Random FXO line" check box is checked, then by dialing a number specified in the text field "Random External Call Access" the User can get the first not busy FXO line.
- If the "Enable Remote Connection" check box is checked, then by dialing a number specified in the text field "Remote Connection Access" the User can get their "own" FXO line. The "own" FXO line is one where its "Destination Phone Number" is the phone number of the Caller.
- If the "FXO->FXS call after second ring" check box is checked, then FXO->FXS call will be started only after the second ring signal is detected. The option can be useful if the User starts to get "junk" calls from the PBX (perhaps as result of voice mail notification).
- If the "Enable Voice Mail" check box is checked, then for FXO-to-FXS calls the phone number of the source FXO gateway will be dialed as the FXS answers. It's used when it is important for the FXS endpoint to know the source gateway of the call.
- If the "Ignore Flash signal" check box is checked, then all flash signals will be ignored. In such a case, transferred and waiting calls are not possible. When the check box is unchecked, the flash signal will be processed by Call Manager, so waiting and transfer calls will be possible.
- When the "Transfer flash to PBX" check box is checked, for FXS <-> FXO calls the flash signal will automatically be transferred to the PBX that the FXO endpoint is connected to.

E.5.9 Scripts

The script screen is only used for AudioCodes internal testing. By default, this screen is disabled.

E.6 Previous Release Notes

E.6.1 Call Manager 2.4 Release Notes

- Added support of signaling for trunk gateways and cards. Signaling is supported over proprietary TPNCP protocol.
- New gateway/endpoint type is added in the GUI – E1/T1 type (for trunk gateways).
- New option via the GUI to add Trunks for E1/T1 gateways.
- Special E1/T1 gateways configuration is added (transport type, protocol, etc.).
- Gateway's prefix must now be two digits.
- Alarm statuses of the E1/T1 gateways can be viewed via the "Gateways & Endpoints Configuration" window.
- "Trunks Monitoring" task is added to manage trunk statuses.
- New call types are possible: FXS->E1/T1, E1/T1->FXS, E1/T1->E1/T1 (ISDN or CAS protocols are used for trunk gateways).
- In the "Statistics" window, "Errors Number" group, a new sub-group "TPNCP Module" was added.

E.6.2 Earlier Release Notes

- Deletion of all existing connections when exiting or deactivating the program. The User can disable the feature via the "Configurations" window ("General" tab).
- Error and warning messages of the logger are **also** saved in a different file. The parameters of the file can be managed via the "Configurations" window ("Logger" tab).
- Filtering logger messages is possible. Filtering can be applied to the "Debug" window or log file or both. The filter can be configured via the "Configurations" window ("Logger" tab).
- New type of log messages added – Warnings. Such messages are usually generated when a Call fails or there are communication problems.
- In the "Gateways & Endpoints Configuration" window, an option was added to select a group of gateways/endpoints. A number of operations (such as removing, disabling, enabling) can be applied to the group.

- In the "Gateway Properties" window, an option was added to change the Gateway's IP address (only when Call Manager is not active).
- In the "Endpoints Properties" window, a new field "Description" was added. It's a regular text field for administrator notes concerning the endpoint.
- FXS *immediate calls* are added. *Immediate calls* are regular calls where the phone number is dialed automatically when lifting the handset. The phone number to be dialed should be defined in the "Endpoint Properties" window.
- Gateway's phone number is added. The phone number is used when the User wants to select a random line of the specified Gateway. Note that the chosen endpoint's type will be the same type as the parent gateway.
- FXO to FXO calls are added.
- In the "Statistics" window, "Failed Calls" group, a new sub-group "Timeout Exceeded" was added.
- Added support of waiting calls for FXS to FXS calls
- Added support of transferred calls for all types of calls.
- New configuration parameter "Transfer flash to PBX" was added to the "Configurations" window ("Telephony" tab). The parameter is used to define the program operation when a "Flash" signal is received on the FXS line.
- Added option to back up and restore the program's database. The option is available from the "File" menu in the main window.
- Digit map is extended to get "xx#".
- Gateway's type can be changed in run-time.
- The "Audit task" of the program is changed. Different levels of monitoring are now possible:
  - Endpoint monitoring;
  - Gateway monitoring;
  - Call monitoring;
- For FXO to FXO calls, it's possible to disconnect a call when silence is detected in one or both directions.
- A special feature has been added in FXO to FXS calls. The prefix phone number of the source FXO gateway will be dialed when the destination FXS answers.

- In FXO to FXS calls, the scenario for busy destination is changed. According to the new scenario, the incoming call from PBX will not be answered and the caller will get a ring back tone generated by the PBX. (According to the old scenario, the incoming call was answered and the caller used to get a busy signal.)

Appendix F - GUI Demo Application

This section describes the Installation & Demontration of the GUI demo Application.

F.1 Overview of the GUI Demo Application

AudioCodes' Customers can use AudioCodes' GUI Demo Application Graphical User Interface to:

- Get started with an AudioCodes product immediately after installation.
- Test AudioCodes' TrunkPack product features and benefits.
- Test AudioCodes' coder compression, PSTN capability, conferencing capabilities, and capabilities generally.

Note that although AudioCodes' TrunkPack Demo Application GUI is mostly used with boards/modules, it can be used with media gateways. In this section, references to boards/modules apply to media gateways as well.

F.1.1 Software Installation

Before starting to use AudioCodes' GUI Demo Application, the following .dll files should be placed in the default directory (the files can be found at..\TPVoIPDemo\Dlls on the board companion cd).

- dlreg.dll
- libdxxmt.dll
- libsrlmt.dll
- ndi.dll
- urlmon.dll
- winstrm.dll

F.1.2 WinDriver Installation on Windows NT and 2000

WinDriver is the TrunkPack board PCI driver and it must be installed in order to access the board. (Note that installation of the WinDriver does not apply to media gateways.)

➢ To install the TrunkPack board PCI driver, take the next 3 steps:

1. Copy the .\windrvr\nt\redist\windrvr.sys file to c:\winnt\system32\drivers.

2. Add WinDriver to the list of device drivers that the operating system loads on boot. This is done by clicking Start/Run and entering:

".\WinDrvr\util\wdreg.exe install" in the textbox and clicking 'OK'.

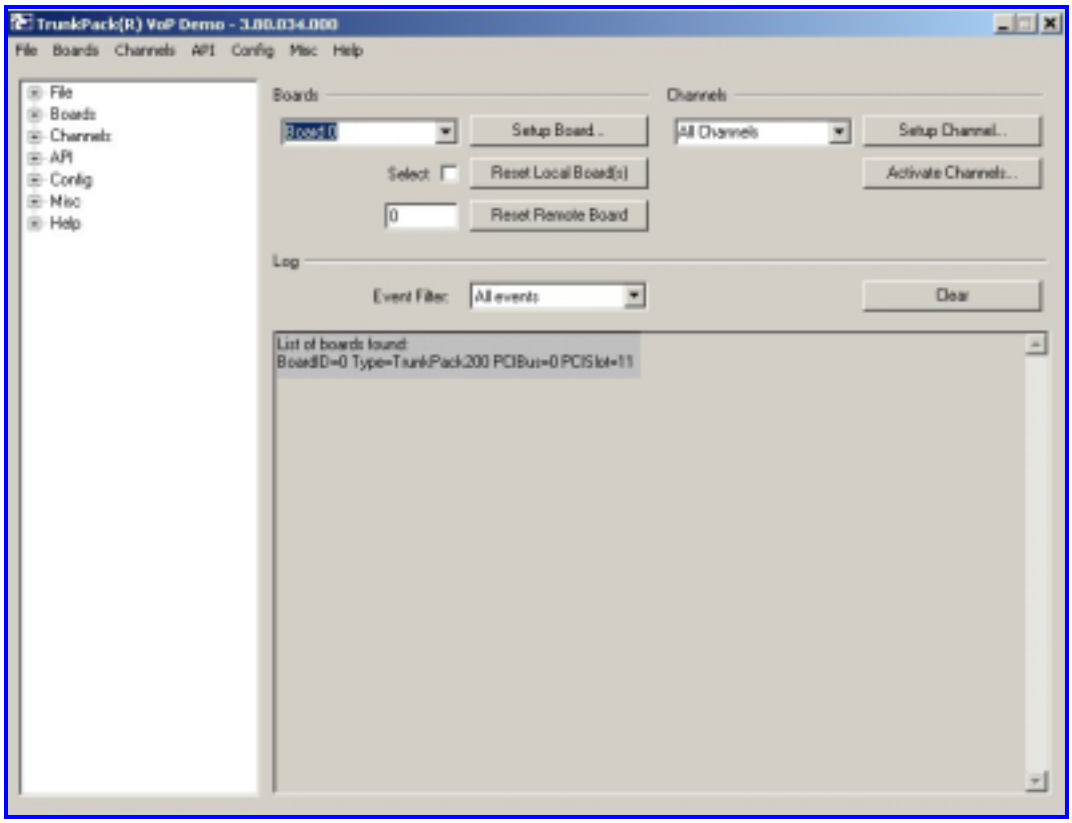
3. Reboot the computer to start the WinDriver.

F.1.3 Getting Started

➢ To get started, take the next 3 steps:

1. Execute Gui.exe; the TrunkPack VoP Demo screen is displayed:

Figure F-1: TrunkPack VoP Demo Main Screen



2. In the main pane under 'List of Boards Found', verify that AudioCodes' Demo Application has recognized the board correctly.

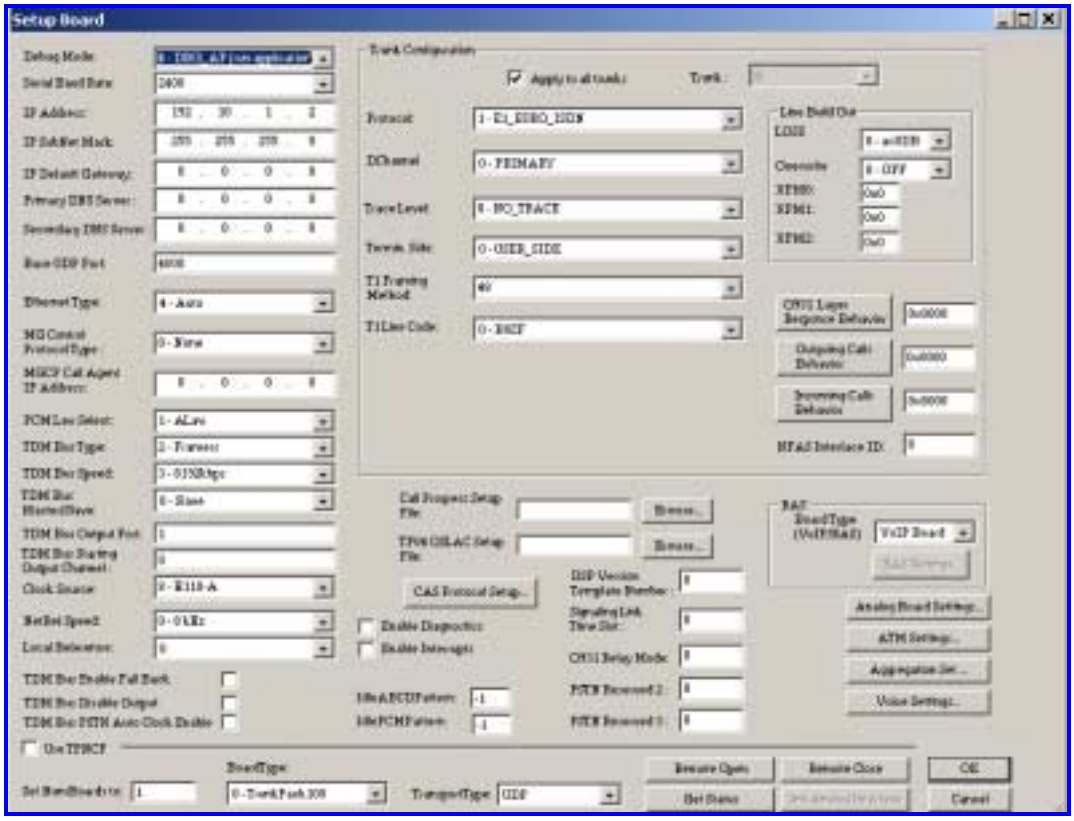
3. In the case that the board is not recognized by the application, reinstall the WinDriver.

F.1.4 Setting up the Board

➢ To set up the board:

- Click the "Setup Board " button; the Setup Board screen is displayed.

Figure F-2: Setup Board



F.1.5 Debugging

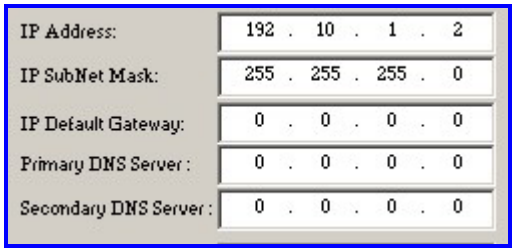
Use Debug Mode for internal debugging only. Make sure that you select Normal operation, as described below:

Figure F-3: Debug Mode

F.1.6 Network Setting

The following parameters set the board's IP address:

Figure F-4: Setting the Board's IP Address


IP Address:
192 . 10 . 1 . 2
IP SubNet Mask:
255 . 255 . 255 . 0
IP Default Gateway:
0 . 0 . 0 . 0
Primary DNS Server :
0 . 0 . 0 . 0
Secondary DNS Server :
0 . 0 . 0 . 0

F.1.7 Base UDP Port Setting

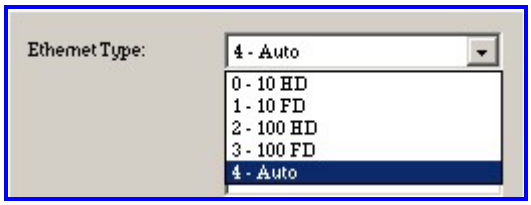
Base UDP port is a number that will be used by the first channel. Every channel should have its own unique port number.

Figure F-5: Base UDP Port Setting

F.1.8 Network Interface Configuration

Normally, the network interface should be set to (4 - Auto Negation). When you need to force a configuration, select one of the following configuration options:

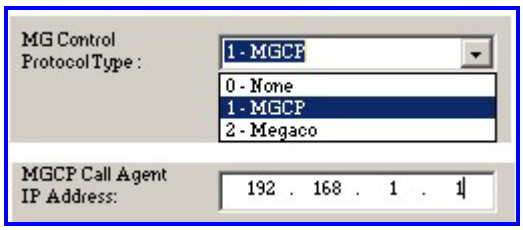
Figure F-6: Ethernet Type (Network Interface) Configuration



F.1.9 MGCP/MEGACO Setting

- Select the desired control protocol: MGCP or MEGACO.
- Specify the Call Manger's IP address.

Figure F-7: MGCP/MEGACO Setting



F.1.10 Selecting PCM Law

Select the PCM law. Usually, A-law is selected for E1 lines and μ-law for T1.

Figure F-8: Selecting PCM Law

F.1.11 Selecting TDM Bus Type

By selecting a TDM bus type, you set the primary PCM interface.

Framers - the board will use the E1/T1 interface

MVIP - MVIP bus interface

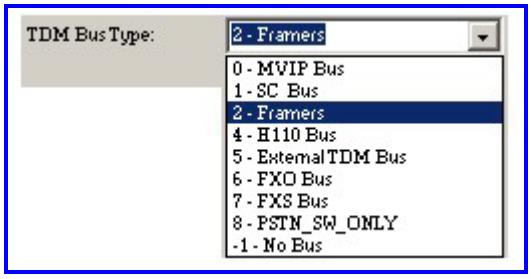
SC - SC bus interface

H.110 - H.110 bus interface

FXS Bus - The analog interface will be FXS. This means, standard phone.

FXO Bus - The analog interface will be FXO. This means, connected directly to the PABX.

Figure F-9: Selecting TDM Bus Type



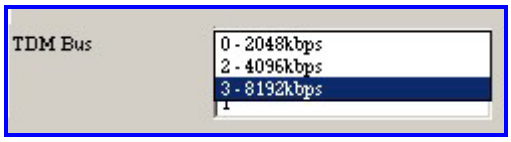
F.1.12 Selecting TDM Bus Speed

MVIP bus - 2048

SC bus - 4096

H.110 bus - 8192

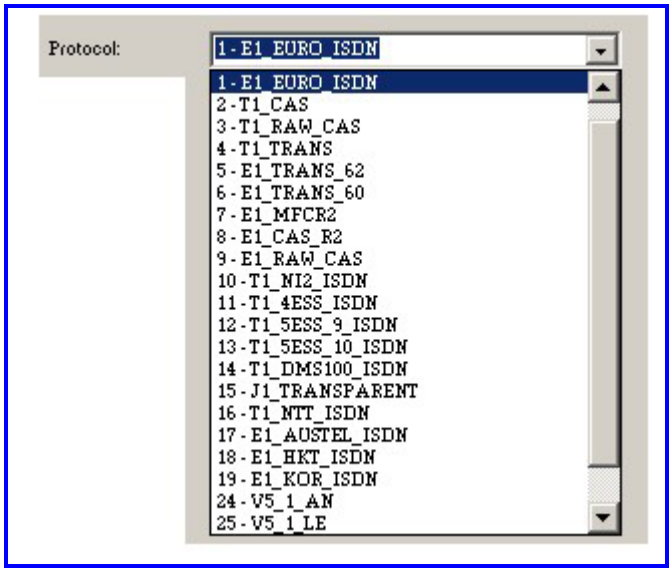
Figure F-10: Selecting TDM Bus Speed



F.1.13 Trunk Configuration

- Select the correct trunk configuration.
- Note that CAS and MFCR2 protocol needs an external state machine file.

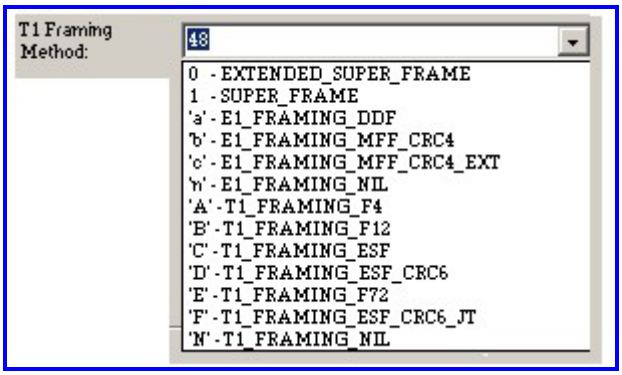
Figure F-11: Trunk Configuration - Protocol



F.1.14 T1 Framing Method

- Select the correct T1 framing method. Usually, it will be Extended Super Frame.

Figure F-12: T1 Framing Method

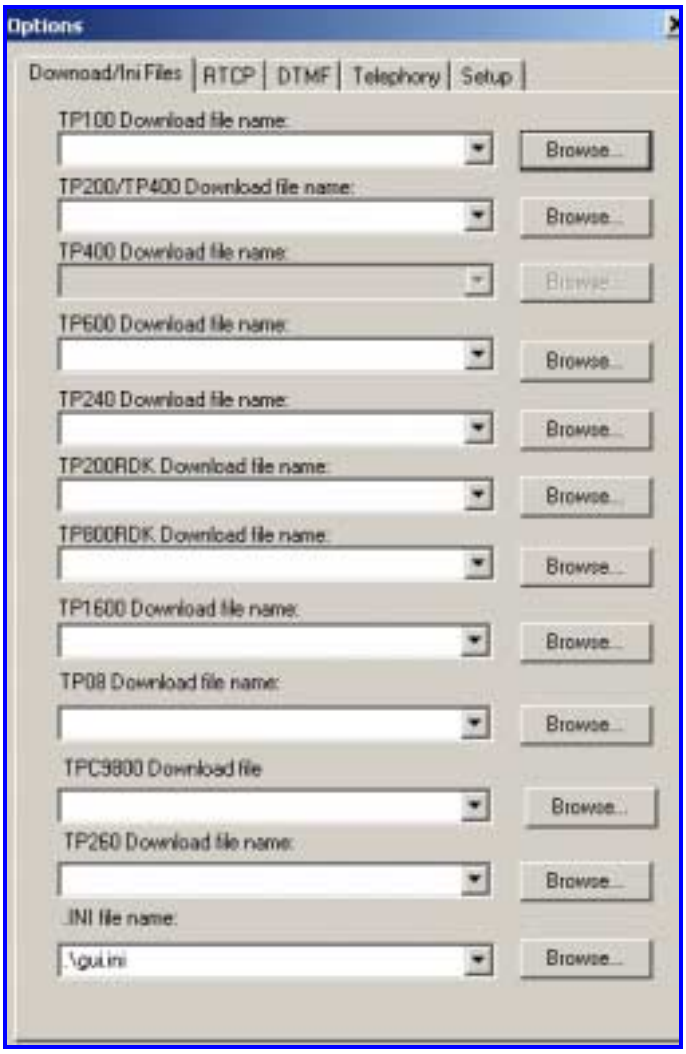


F.1.15 Downloading the Board's Firmware

➢ To download the board firmware:

1. In the TrunkPack VoP main screen, select 'Options' from the 'Config' menu; the download files are usually located at ...\\vouplib\downlaod
2. To download the file and configuration parameters, press "Reset".

Figure F-13: Downloading Firmware

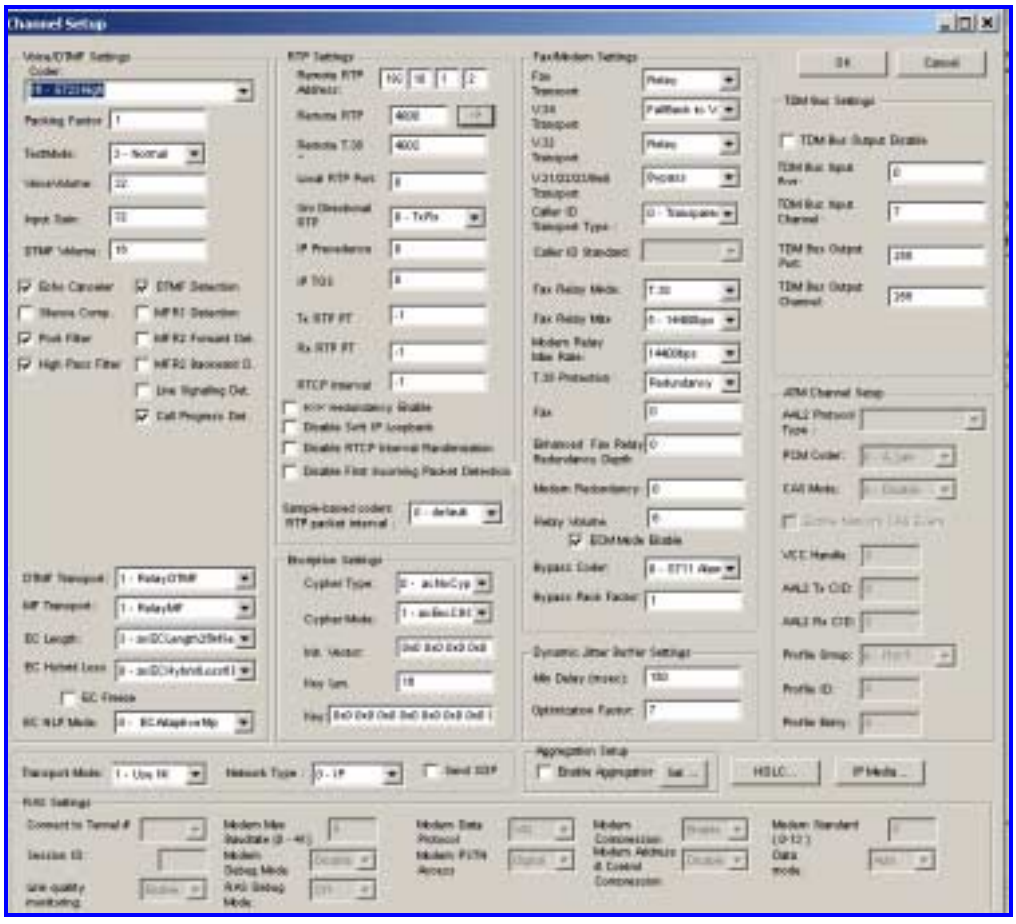


F.1.16 Setting up Channels

To set up channel parameters, select the desired channel from the channel list box. You can set the channels one by one, or set them all with the "All Channels" option.

Figure F-14: Setting Up Channels

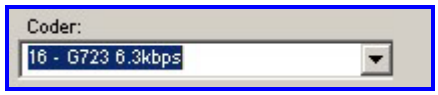
Figure F-15: Channel Setup



F.1.17 Voice Coder Setup

To select the voice coder, select one from the drop-down list. Note that every voice coder can be set independently.

Figure F-16: Voice Coder Setup



F.1.18 RTP M - Factor

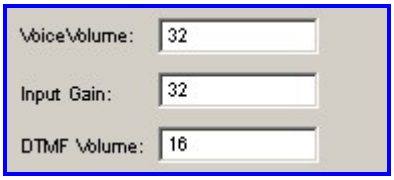
Select the RTP multiplication factor.

Figure F-17: RTP Multiplication Factor

F.1.19 Voice Volume

The units are in db when 32 is equal to 0 db.

Figure F-18: Voice Volume

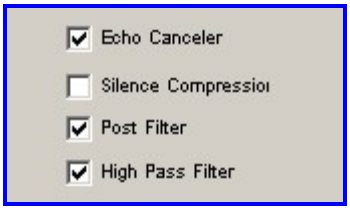


F.1.20 Filters and Echo Cancelation

Beside the Echo Canceler, you will need to set the Echo Canceler length.

Pre and Post filter are mandatory for the G 723.1 voice coder.

Figure F-19: Echo Cancelation & Filters



F.1.21 In-Band Signaling Detectors

The next in band signaling is available. Do not select more then two at the same time.

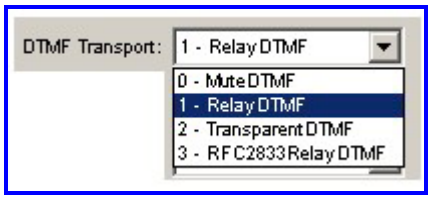
Figure F-20: In-Band Signaling Detection

F.1.22 DTMF Transport Type

Select the DTMF transport type.

- Mute DTMF
- Relay DTMF use a propriety DTMF relay (Payload type 100)
- Transparent DTMF
- RFC 2833 Relay DTMF, relay DTMF according to the RFC

Figure F-21: DTMF Transport Type



F.1.23 RTP Settings

This sets the remote RTP address. To loop the RTP stream, use the same IP address as the board address.

The RTP address is the same as the UDP port address

RTP = Base UDP port

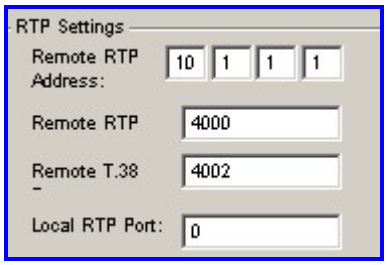
RTCP = Base UDP port + 1

T38 = Base UDP port + 2

The interval between each channel is always 10.

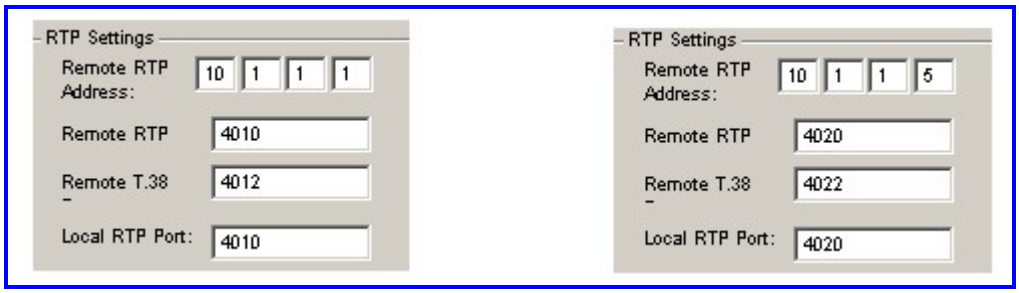
Local RTP port is the port number that the local channel is waiting on for the incoming RTP stream.

Figure F-22: RTP Settings



For example, in order to set a Voice over IP connection between channel 1 and 2 on different boards:

Local Board channel 1 **Remote Board channel 2**



F.1.24 TDM Bus Settings

When using a TDM bus interface such as SC/MVIP/H.110, the input and output of the TDM bus must be set to the correct values.

(Stream: Time Slot)

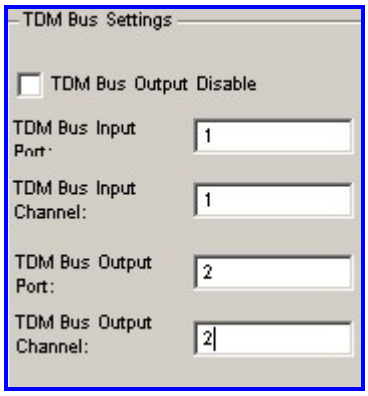
TDM Bus input port is equal to the input stream number.

TDM Bus input channel is equal to the input Time slot.

TDM Bus output port is equal to the output stream number.

TDM Bus output channel is equal to the output Time slot.

Figure F-23: TDM Bus Settings



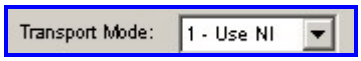
F.1.25 PCI or NI Use

NI means that the network interface will be used for packet transition.

PCI means that the PCI interface will be used for packet transition.

Note that to play or record, the PCI interface should be used for that channel.

Figure F-24: PCI or NI for Packet Transition



F.1.26 Jitter Buffer Optimizations

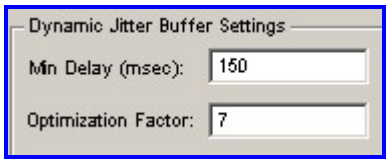
Jitter Buffer 'Minimum Delay (msec)' sets the minimum delay of the dynamic jitter buffer. The jitter buffer is very important in the case of a jittered network. It can dramatically improve voice quality.

The correct values for this field are between 0 - 300 msec.

The 'Optimization Factor' defines how fast the buffer will adjust to the accurate jitter. The correct values for the Optimization Factor' field are from 0 – 12.

To disable the Jitter Buffer, set both parameters to 0.

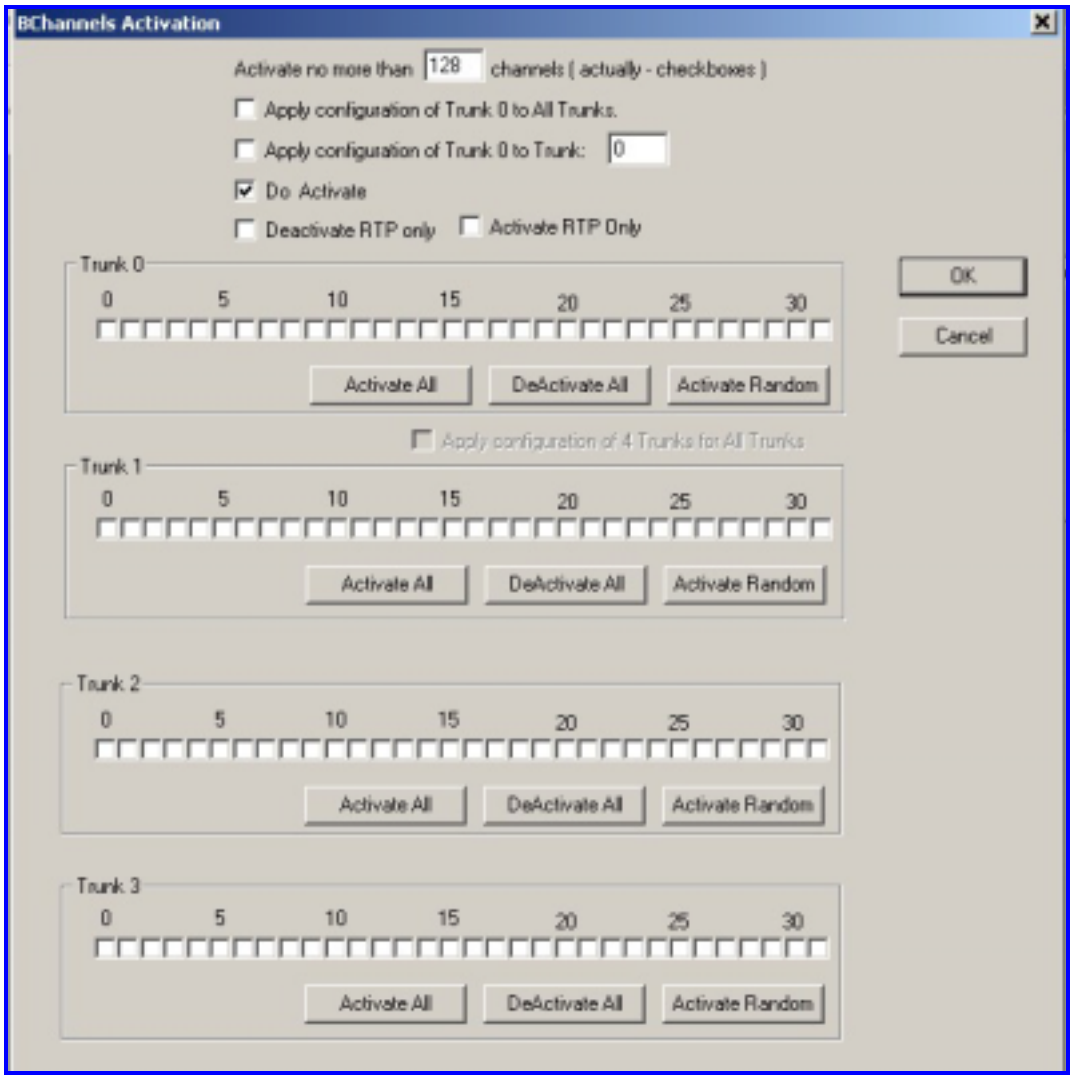
Figure F-25: Jitter Buffer Optimizations



F.1.27 Channel Activation

- The channels are allocated according to the PSTN interface.
- Channel 0 and 16 cannot be activated.
- To activate the channel without RTP, uncheck the 'Do Activate' check box.

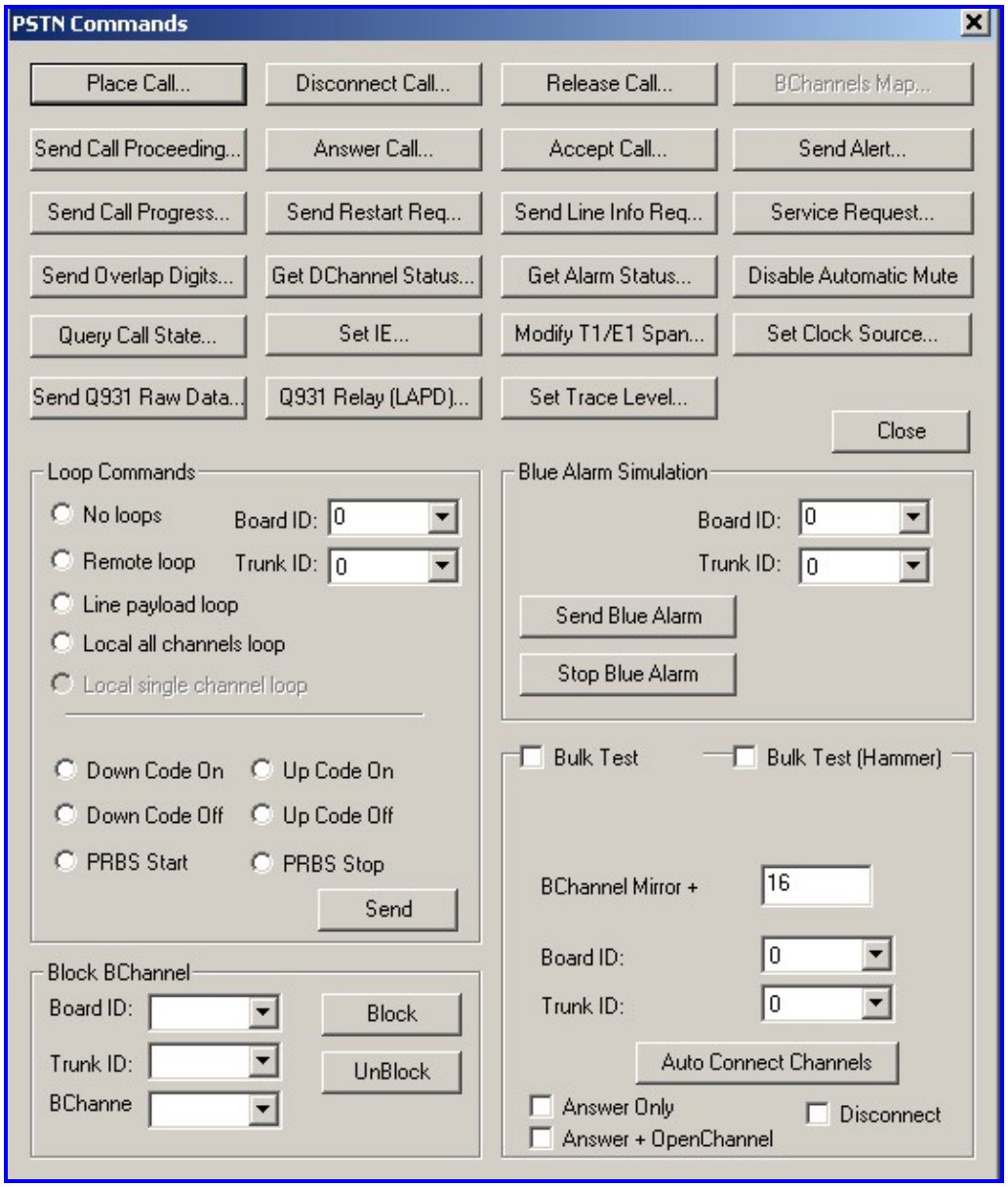
Figure F-26: Channel Activation



F.1.28 PSTN Handling

The PSTN interface can be handled using the API PSTN section.

Figure F-27: PSTN Commands

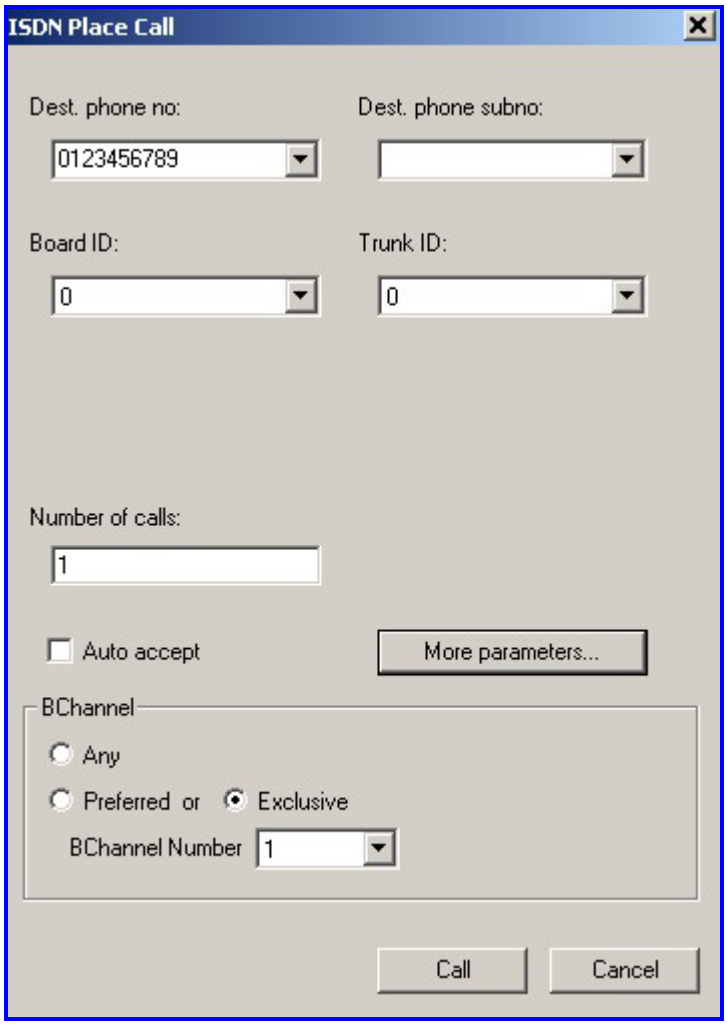


F.1.29 Creating a New Call

To create a new call, use the 'Call' button in the Place Call screen.

Remember to select the correct trunk and B Channel.

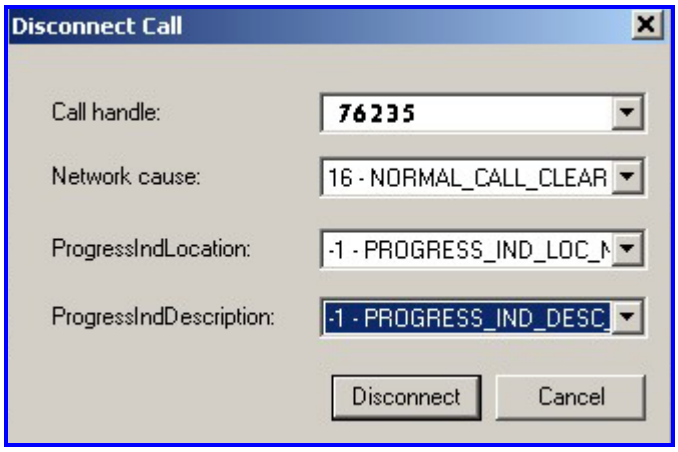
Figure F-28: Placing a Call



F.1.30 Disconnecting a Call

To disconnect a call, the correct call handle must be used. The call handle will be displayed with the Call_Connected event.

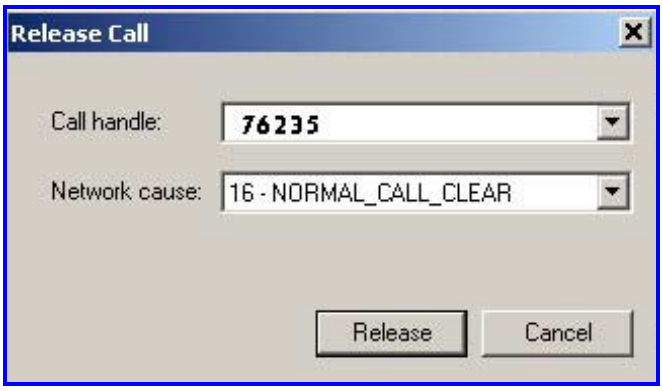
Figure F-29: Disconnecting a Call



F.1.31 Releasing a Call

To release the call, the correct Call handle must be used.

Figure F-30: Releasing a Call

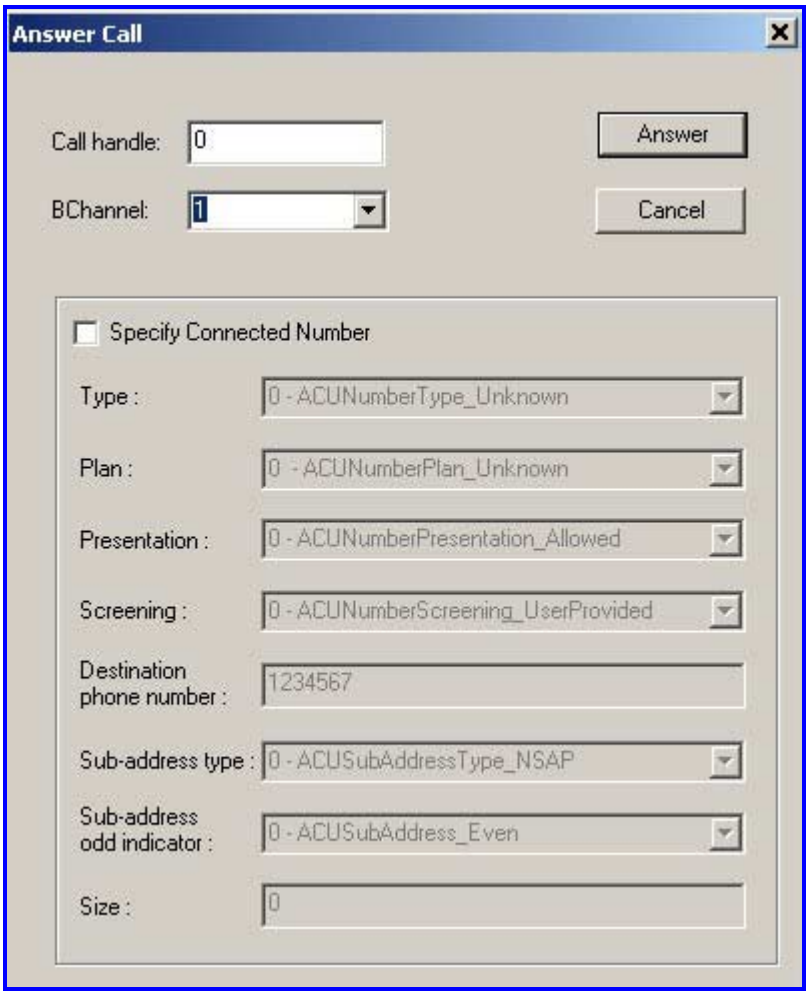


F.1.32 Answer Call

Answer Call is only valid for non-MFCR2 incoming calls.

To answer a call, the correct Call handle must be used. The Call handle is received during the seizure event.

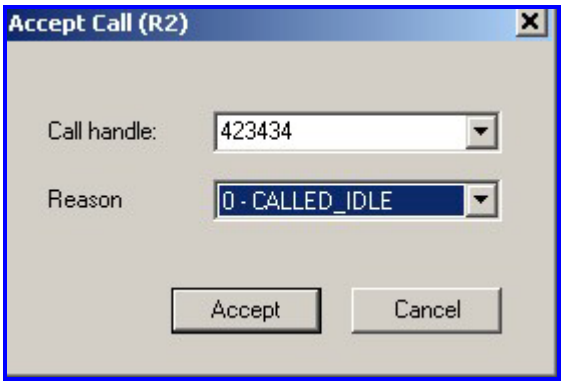
Figure F-31: Answer Call



F.1.33 Accepting a Call

The next command is relevant only for MFCR2.

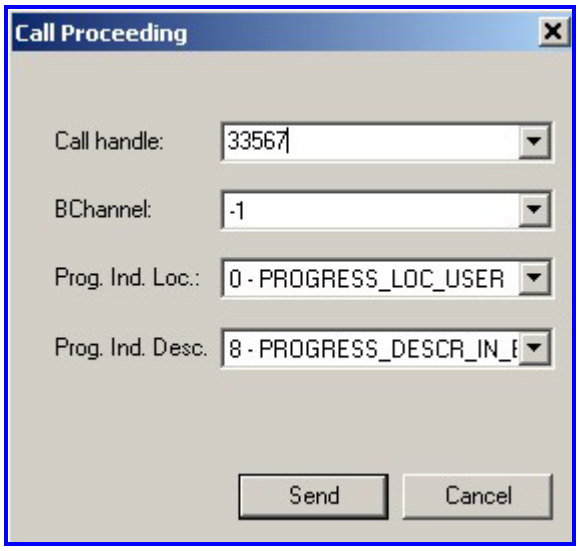
Figure F-32: Accept Call



F.1.34 Call Proceeding

Use this command to signal proceedings to the remote side. Call handle must be used.

Figure F-33: Call Proceeding

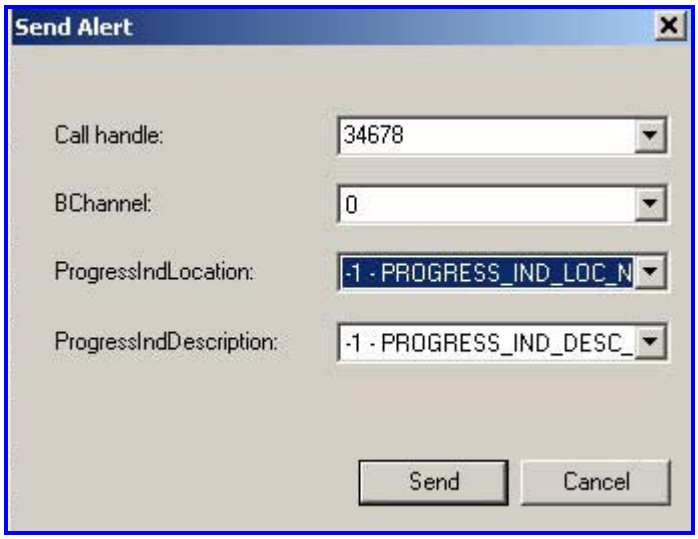


F.1.35 Call Alerting

Use this command to alert the remote side after detecting an incoming call.

Call handle must be used.

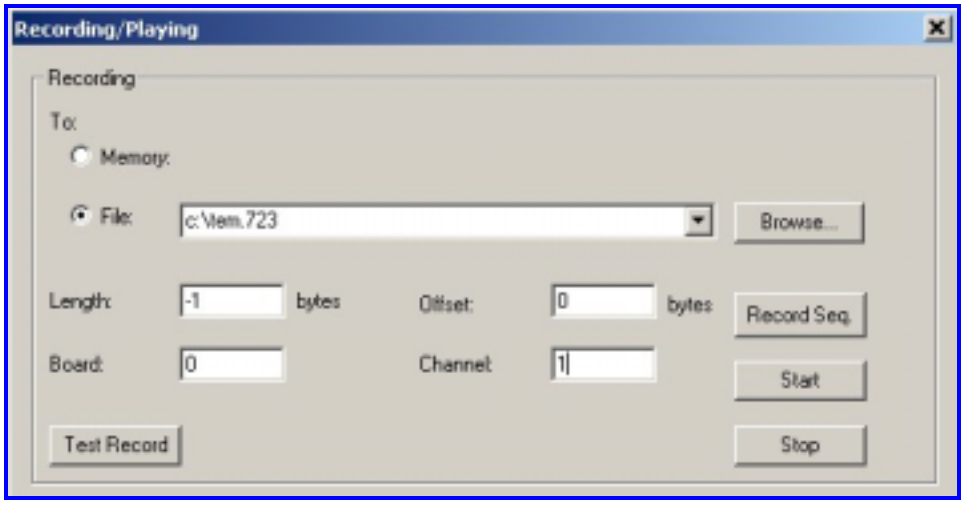
Figure F-34: Call Alerting (Send Alert)



F.1.36 Play and Record

To play or record a file to the PSTN side, Channel must be active and the "Packets transfer mode" set to PCI. Length can be predefined or set to –1 for continuous recording or playing.

Figure F-35: Play and Record

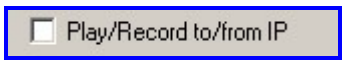


F.1.37 Play and Record from the IP Side

To play or record using the VoIP interface:

- Channel must be open in NI mode and the RTP is active
- The 'PlayRecord to/from IP' check box must be checked

Figure F-36: Play/Record to/from IP

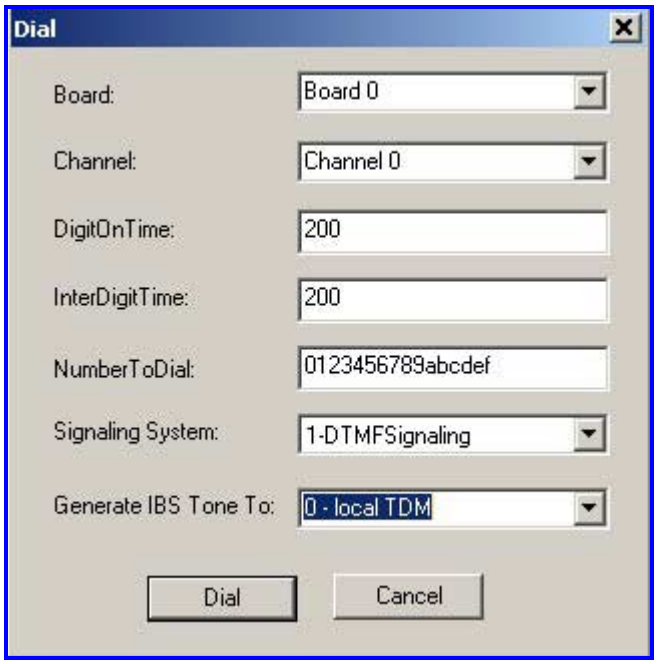


F.1.38 DTMF Playing

➢ To play DTMF

- The channel must be active.
- Digit On Time sets the on time of the DTMF digit.
- Inter Digit Time sets the time interval between each DTMF digit.
- Signaling System can be DTMF or MF.
- 'Generate IBS Tone To' can be the local PSTN or the packet network side, according to RFC 2833.

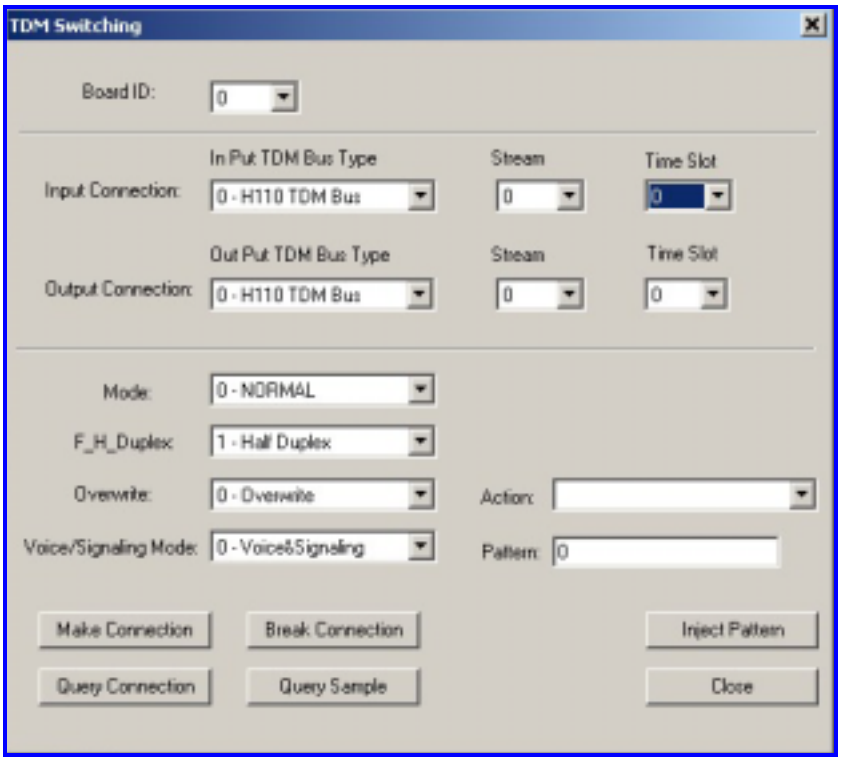
Figure F-37: Dial



F.1.39 TDM Local Switching

TDM local switching is a set of commands that allow the User to manipulate the connection between the PSTN bus/TDM bus and the processing DSP. The User can also create a new connection for:

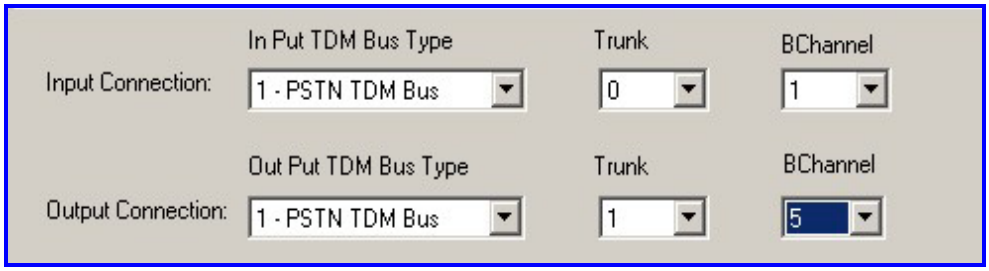
Figure F-38: TDM Switching



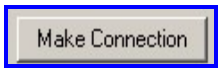
Example: To create the following connection:

Trunk 0 B channel 1 → Trunk 1 B Channel 5

Figure F-39: Connecting Trunk 0 B channel 1 → Trunk 1 B Channel 5



Click the button:



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